Quintum Technologies Tenor
Configuration Guide
For Use with AT&T's
IP Flexible Reach Service

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1 Introduction

This Guide describes the steps for configuring Quintum Tenor AS or Tenor AX to work with AT&T's Flexible Reach Service. The Quintum Tenor AS/AX Multipath Gateway Switch provides Analog telephone access with VoIP capability and multipath switching for redundancy. Tenor Software release P104.12.02 was tested with the AT&T IP Flexible Reach Service.

The Tenor AS/AX is a multipath switch capable of supporting Analog telephones. The Tenor AS/AX uses SIP signaling to complete telephone calls over the AT&T IP network using the Flexible Reach service. Under normal conditions, calls from Analog telephones are converted into VoIP packets by the Tenor AS/AX and sent through the AT&T IP network. The Tenor AS/AX can also use a PSTN connection to bypass the data network (AT&T and Customer) if there is an outage.

References in this Service Guide to "Tenor" refer to either or both of the Tenor AS and Tenor AX.

2 Special Notes

Emergency 911/E911 Services Limitations

While AT&T IP Flexible Reach services support E911/911 calling capabilities in certain circumstances, there are significant limitations on how these capabilities are delivered. Please review the AT&T IP Flexible Reach Service Guide in detail to understand these limitations and restrictions.

Tenor Transfer Feature Must be Turned Off

The Tenor IP transfer feature must be turned off. Quintum and AT&T do not currently have a compatible mechanism for handling IP transfers.

Session Description Protocol in Session Progress and Ringing Messages Must be Turned Off

Session description protocol in the session progress and ringing messages must be turned off in the Tenor.

3 Capabilities Overview

The Quintum Technologies Tenor supports the following capabilities in conjunction with the AT&T Flexible Reach service.

3.1 Calling Scenarios Supported

Inbound Calls to the Quintum Tenor

- Offnet gateway inbound to TENOR
  - Local, Long Distance, International, FAX
- TENOR to TENOR
  - Local, Long Distance, International, FAX
- TDM PBX to TENOR
  - Local, Long Distance, International, FAX

Outbound Calls from the Tenor

- TENOR to PSTN Offnet gateway
  - Local, Long Distance, International, FAX
- TENOR to TENOR
  - Local, Long Distance, International, FAX
- TENOR to TDM PBX
  - Local, Long Distance, International, FAX

3.2 Routing Scenarios Supported

- Failover from Primary to Secondary AT&T IP Border Element
- Failover to PSTN when IP network unavailable or degraded quality[1]

3.3 Codecs Supported

- G.729AB 8.0 Kbps
- G.711 A-law 64Kbps
- G.711 Mu-law 64Kbps

3.4 Features Supported

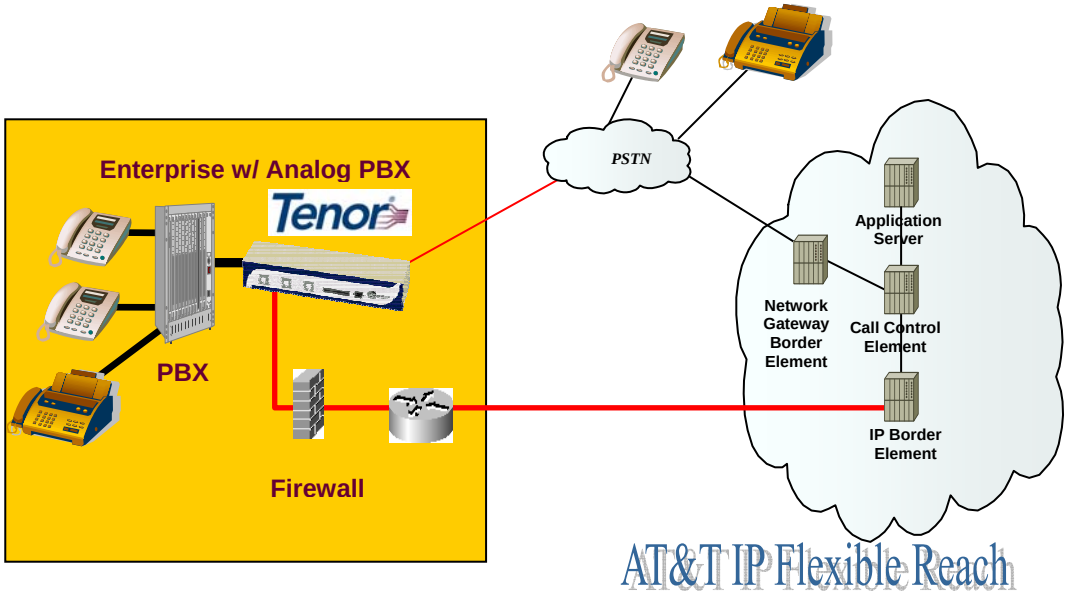
Virtual Telephone Number Support

[1] Quintum patented SelectNet™ Technology monitors the IP network performance for VoIP calls between Tenors. If the performance characteristics become unacceptable—according to packet loss specifications you configure—the *Tenor* will switch the call to the PSTN automatically and transparently.

Calling Name Delivery
Call Hold and Resume
DTMF Relay
In Band DTMF
T38 FAX

4 Configuration Component Overview

This section provides a service overview of the Vendor integration with AT&T IP Flexible Reach.



The customer premises equipment shall consist of the following components.

- Customer PBX with standard 2-wire Analog interface connections for FXO ports.
  Note: The Tenor AS provides a standard Centronics 50 pin male interface (50 pin / 25 pair male Amphenol connector).

  OR

  Customer Analog Phones and/or FAX machines with RJ11 interfaces.
  Note: The Tenor AX provides standard RJ11 interfaces.

- AT&T Managed Router

- Customer optional Firewall

- Quintum Technologies Tenor Analog Gateway (AS or AX).

Tenor Software release P104.12.02 was used when conducting interoperability testing with the AT&T IP Flexible Reach Service.

4.1 Tenor AX Overview

The *Tenor AX* is a high-density VoiP (Voice over Internet Protocol) SIP/H.323 switch that compresses and packetizes voice, fax, and modem data and transmits it over the IP network. The *Tenor AX* gives larger businesses with analog voice infrastructure a means to use Voice over IP (VoIP).

The Tenor's MultiPath architecture enables it to intelligently route calls between the FXS, FXO, and the VoIP network. The *Tenor AX* also routes calls over IP to reduce costs, and then transparently "hop off" to the PSTN, to reach offnet locations. Calls can be routed in any direction between any of the ports.

The unit's plug and play embedded system architecture brings VoIP technology to your network without changing your existing telephony infrastructure. The Customer's network stays as is, and the call type is transparent to the user.

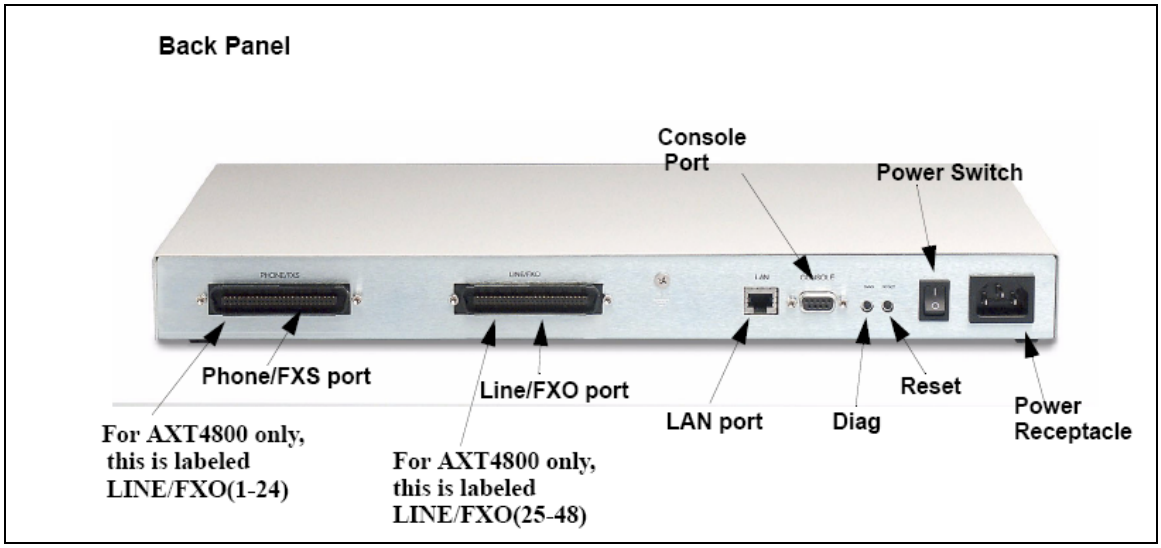


Figure 1 - Tenor AX Back Panel

- **Phone/FXS port -** Provides a 50 Pin Telco connector which supports up to 24 Phone/FXS connections for connecting to the PBX, Keyphone or phones.

- **Line/FXO port -** Provides a 50 Pin Telco connector which supports up to 24 Line/FXO connections for connection to the Central Office (connection to the PSTN).

- **LAN port -** 10/100 Base-T Ethernet port. This port provides an RJ-45 jack for individual connection to a 10/100 Ethernet LAN switch or hub via RJ-45 cable; it is individually configured with a unique IP and MAC address.

The Tenor AX will support 8, 16, 24 or 48 Simultaneous VoIP Calls.

For more details on the Tenor AX, consult with document [1].

4.2 Tenor AS Overview

The *Tenor AS* is a VoIP (Voice over Internet Protocol) H.323/SIP switch that digitizes voice, fax, and modem data and transmits it over the IP network. The *Tenor AS* gives small to medium sized businesses with analog voice infrastructure a means to use Voice over IP (VoIP).

The Tenor's MultiPath architecture enables it to intelligently route calls between the FXS, FXO, and the VoIP network to achieve the best combination of cost and quality. The *Tenor AX* also routes calls over IP to reduce costs, and then transparently "hop off" to the PSTN, to reach offnet locations. Calls can be routed in any direction between any of the ports.

The Tenor can be installed without upgrades to the existing voice or data network. You can install the unit in a home or office environment, without affecting the network infrastructure you already have in place.

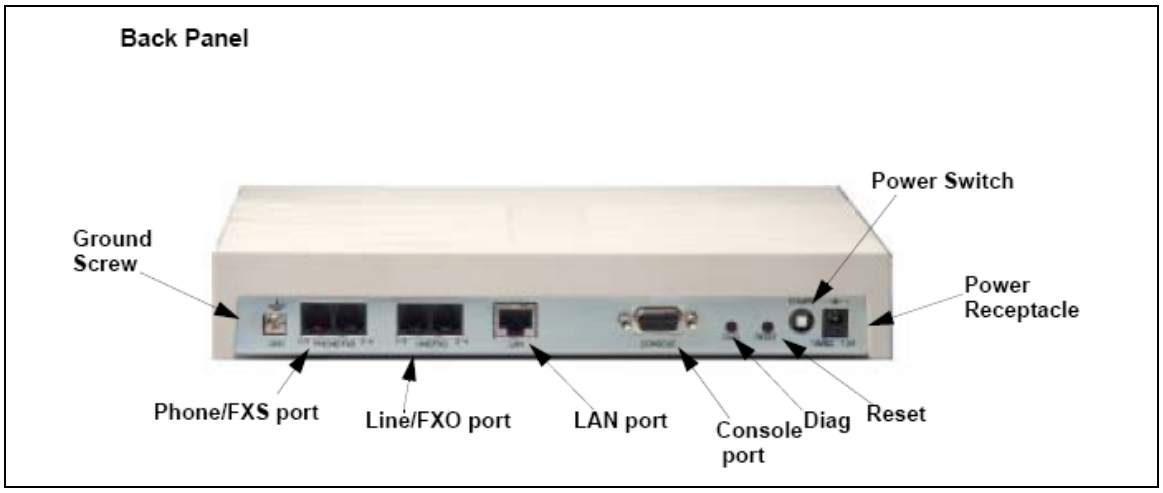


Figure 2 - Tenor AS Back Panel

- **Ground Screw -** An earth ground screw is provided to connect to supplemental earth ground using a Ground Safety Cable, if supplemental ground is needed.

- **Phone/FXS port -** Provides an RJ-11 jack for connection to a PBX, Keyphone or phone.

- **Console port -** This RS-232 connector is used for connection to a PC's serial port via DB-9 serial cable at 38400 BPS 8N1, without flow control.

- **Diag -** Enables you to perform software diagnostic procedures.

- **Reset -** Enables you to reset the system.

- **Power Switch -** Switch to turn power on and off.

- **Power Socket -** Connection port to external power supply block.

The Tenor AS will support 2 or 4 Simultaneous VoIP Calls.
The Tenor AF will support 8 Simultaneous VoIP Calls.

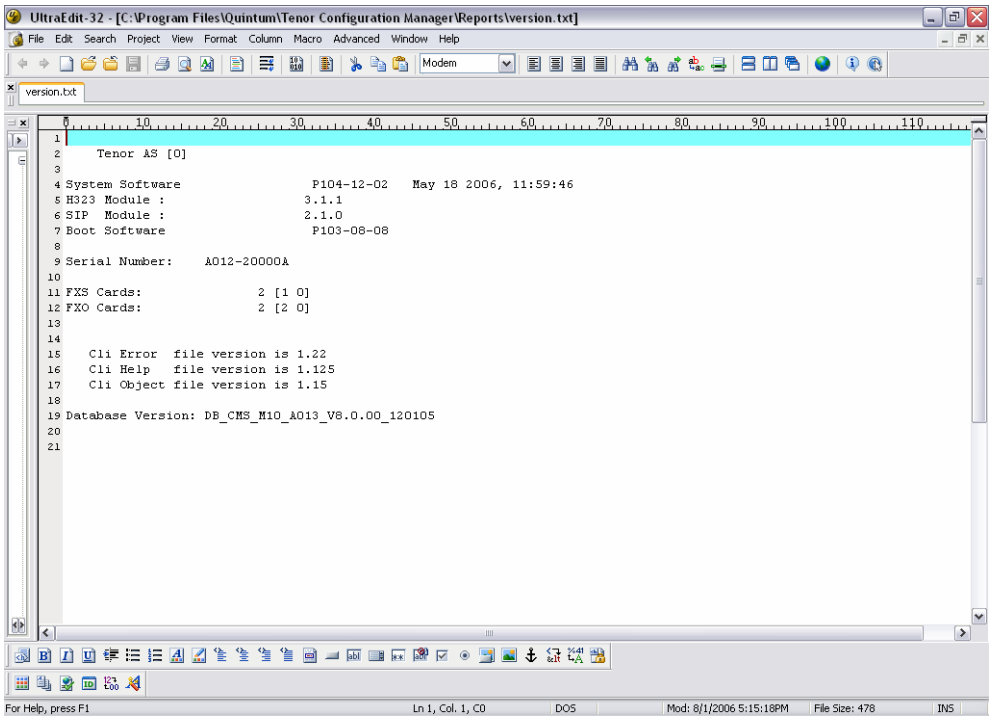
For more details on the Tenor AS, consult with document [2].

5 Configuration Guide

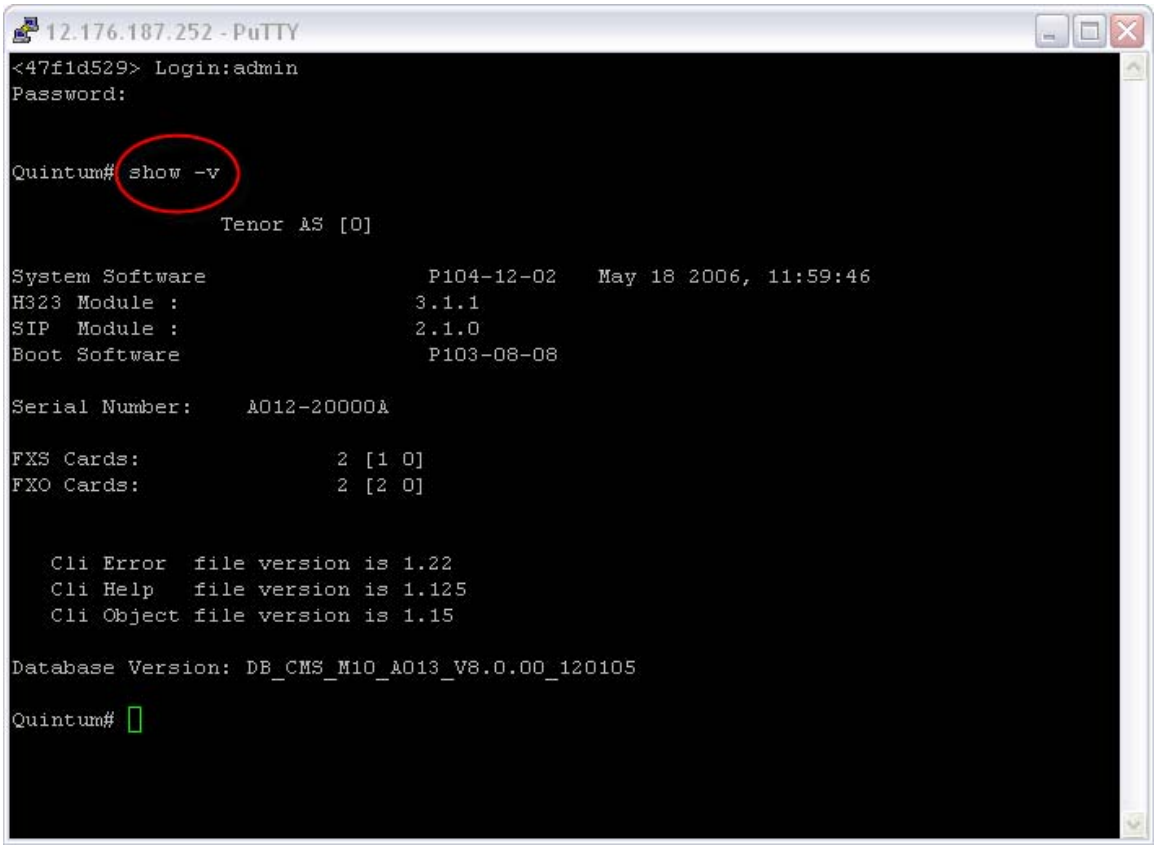
5.1 Tenor Software Version

The version of the Tenor Software can be obtained via the Tenor Configuration Manager GUI or the Command Line Interface (CLI).

From the Configuration Manager View Menu, click on “Tenor Version”.
A text file will open in a new window displaying the Software version as shown below.


UltraEdit-32 - [C:\Program Files\Quintum\Tenor Configuration Manager\Reports\version.txt]
File Edit Search Project View Format Column Macro Advanced Window Help
Modem
version.txt
Tenor AS [0]
System Software P104-12-02 May 18 2006, 11:59:46
H323 Module : 3.1.1
SIP Module : 2.1.0
Boot Software P103-08-08
Serial Number: A012-20000A
FXS Cards: 2 [1 0]
FXO Cards: 2 [2 0]
Cli Error file version is 1.22
Cli Help file version is 1.125
Cli Object file version is 1.15
Database Version: DB_CMS_M10_A013_V8.0.00_120105
For Help, press F1
Ln 1, Col. 1, C0
DOS
Mod: 8/1/2006 5:15:18PM
File Size: 478
INS

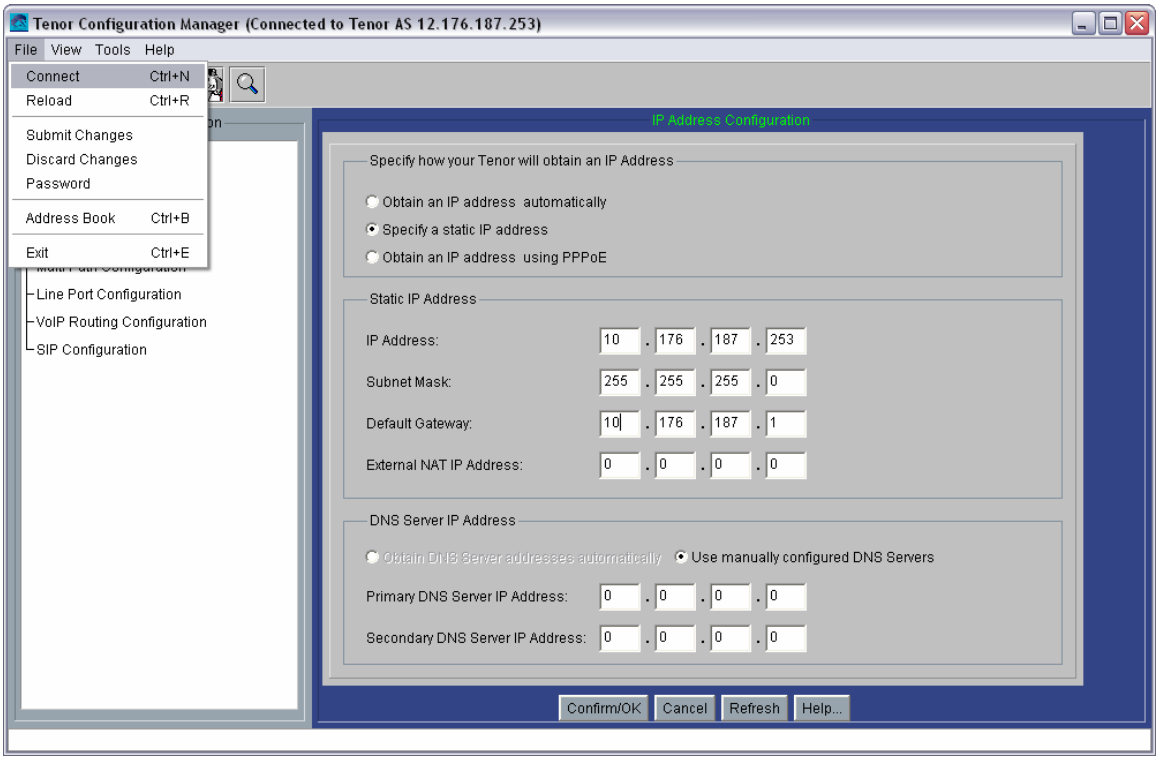
As shown below, the CLI Command to display the Tenor Software version information is “show –v”.

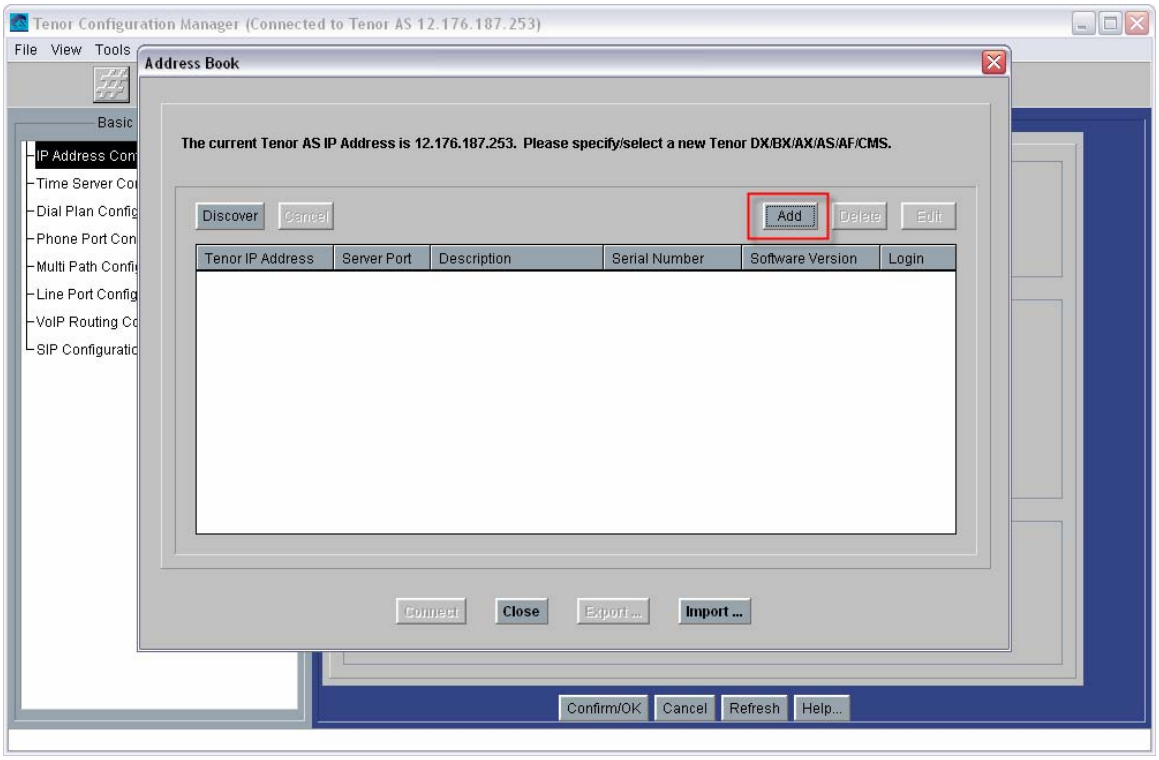


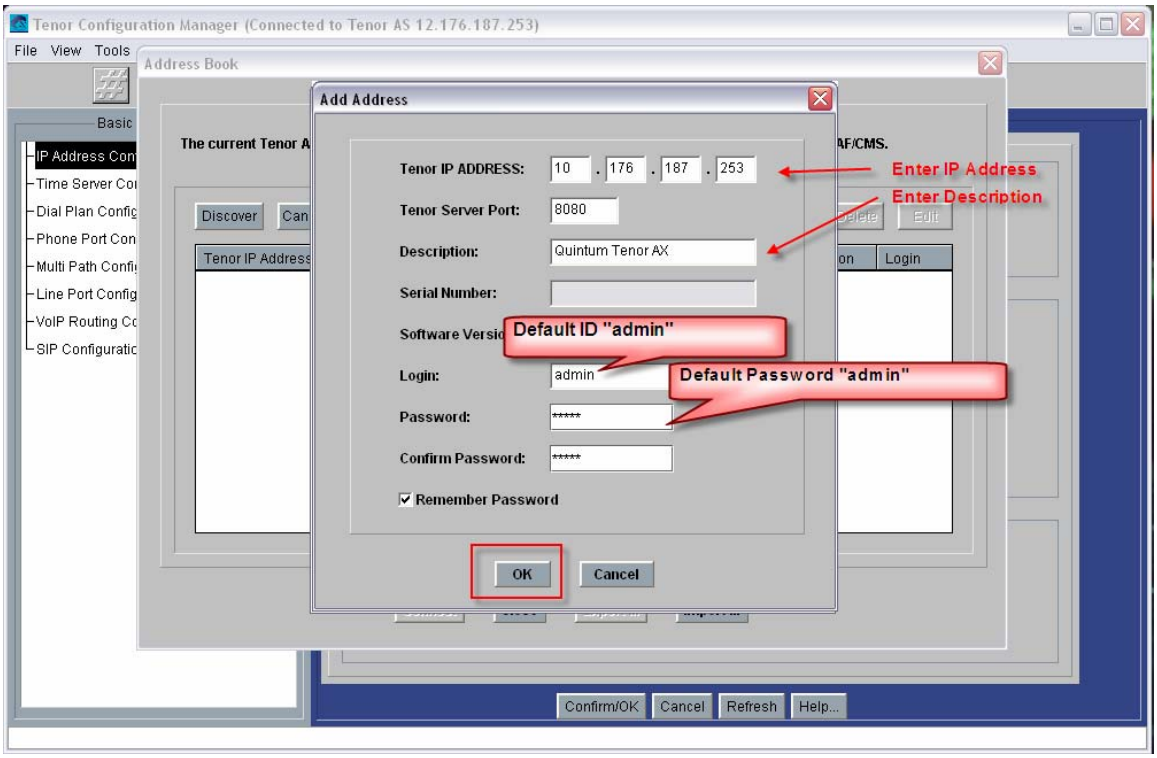
For technical support on the Quintum Tenor AS and Tenor AX, contact Quintum at 877-435-7553, and also refer to www.quintum.com

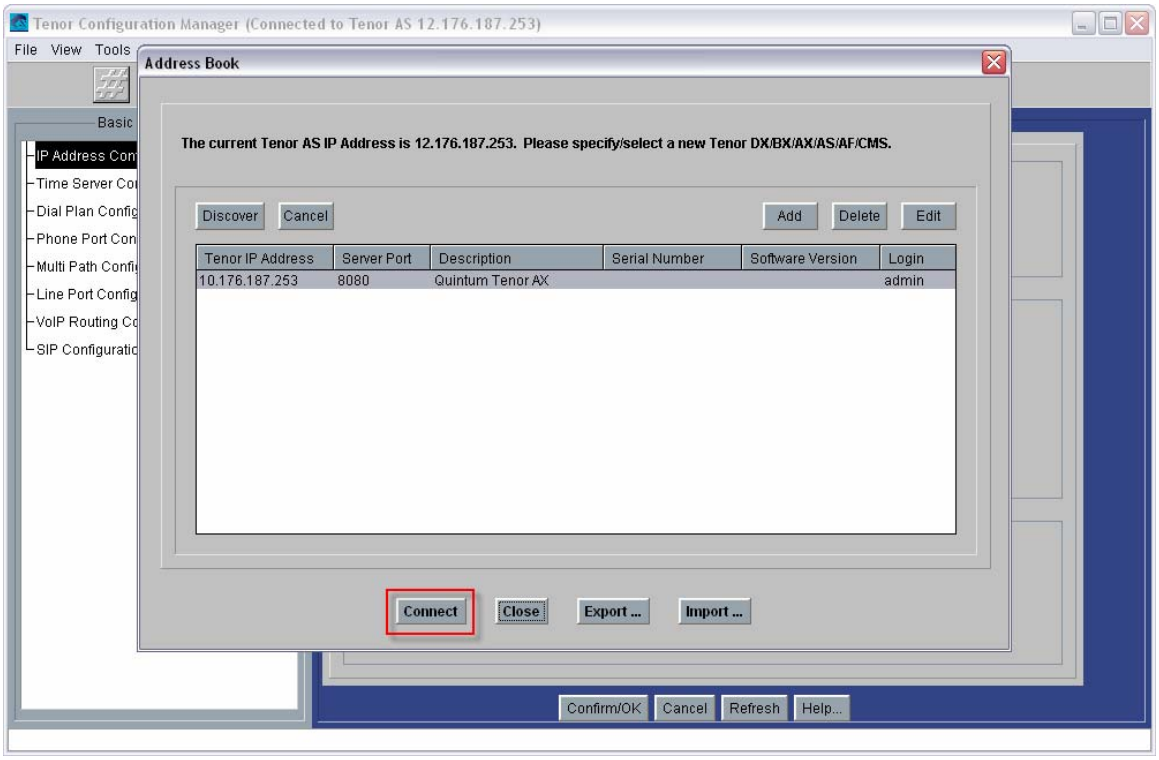
5.2 Standard Configuration

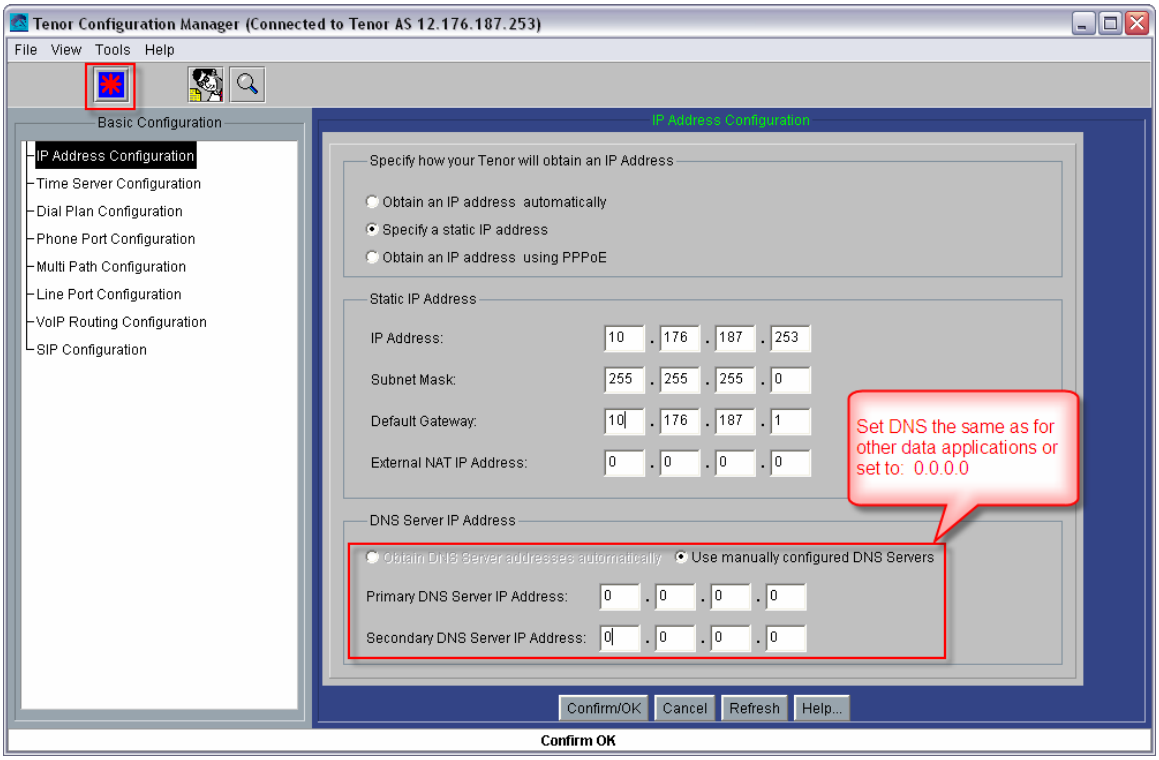
The following steps describe the configuration for the Tenor AS Multipath Gateway Switch verified to work with the AT&T IP Flexible Reach service. Configuration for the Tenor AX is the same as the Tenor AS described below. For detailed information on installing and running Tenor Configuration Manager, consult documents [1], [2] and [3].

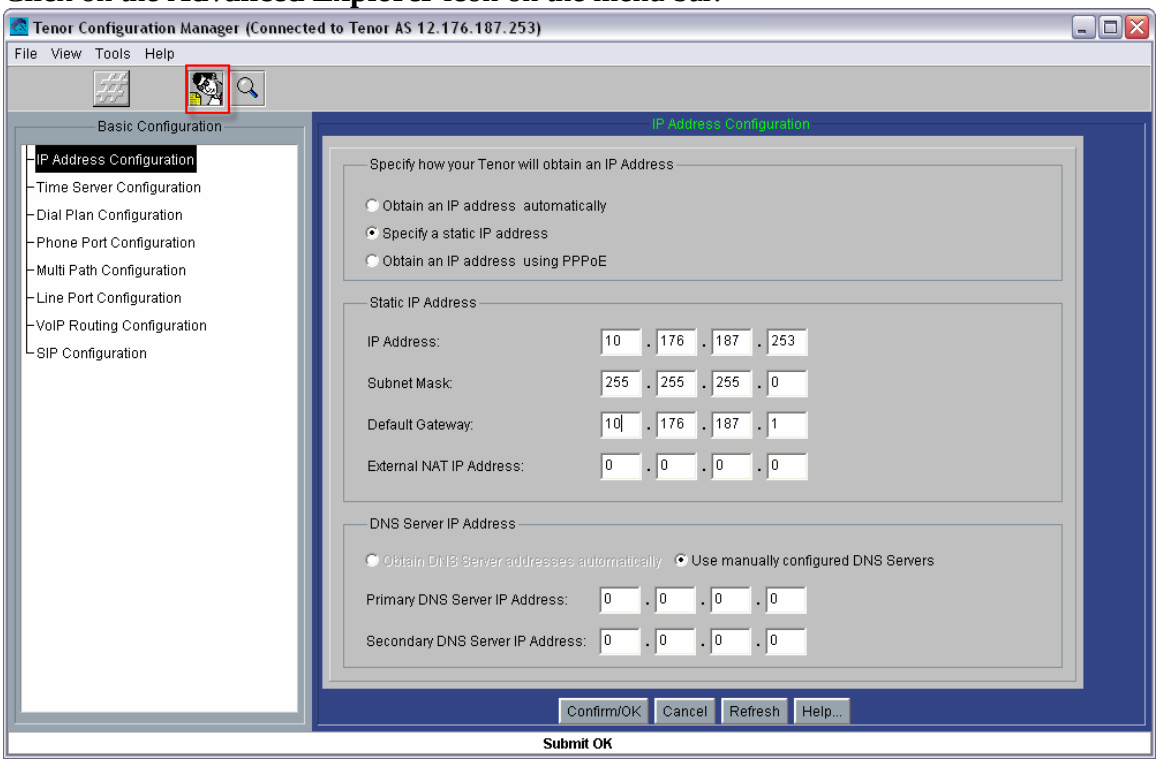
Step	Description
1.	Run the Tenor Configuration Manager. From the File Menu click on **Connect**. 

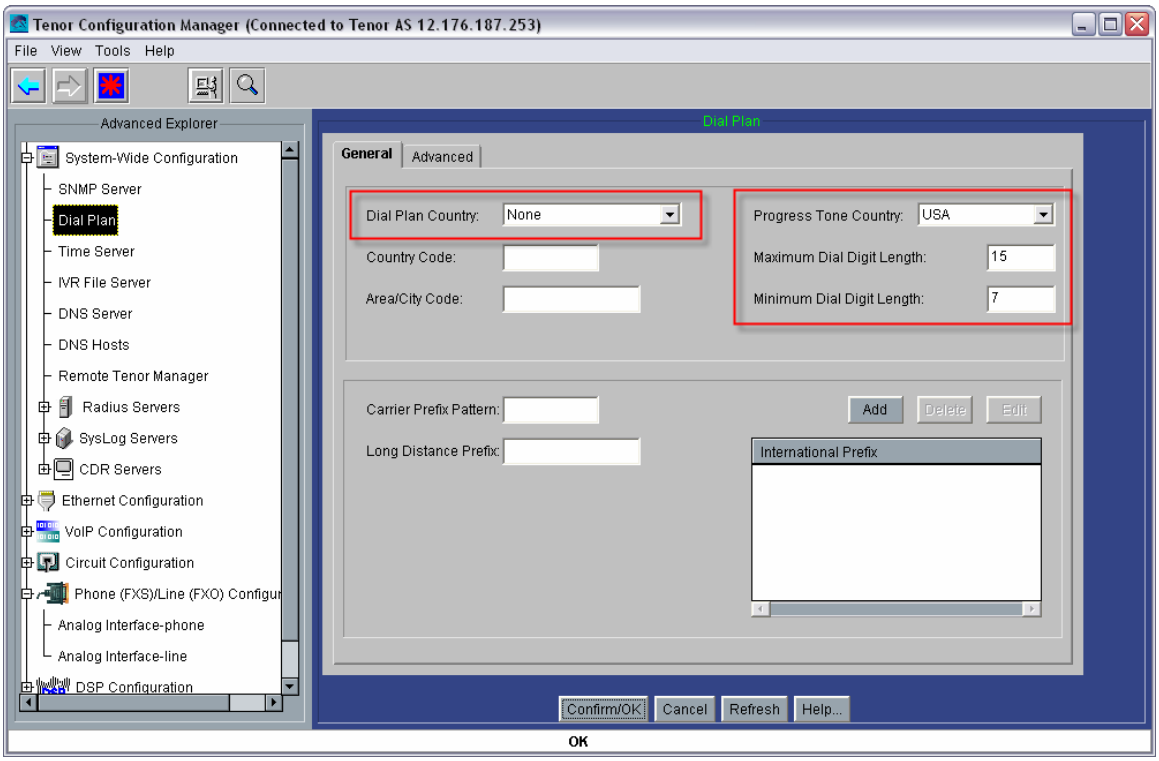
Step	Description
2.	Click on **Add.** 

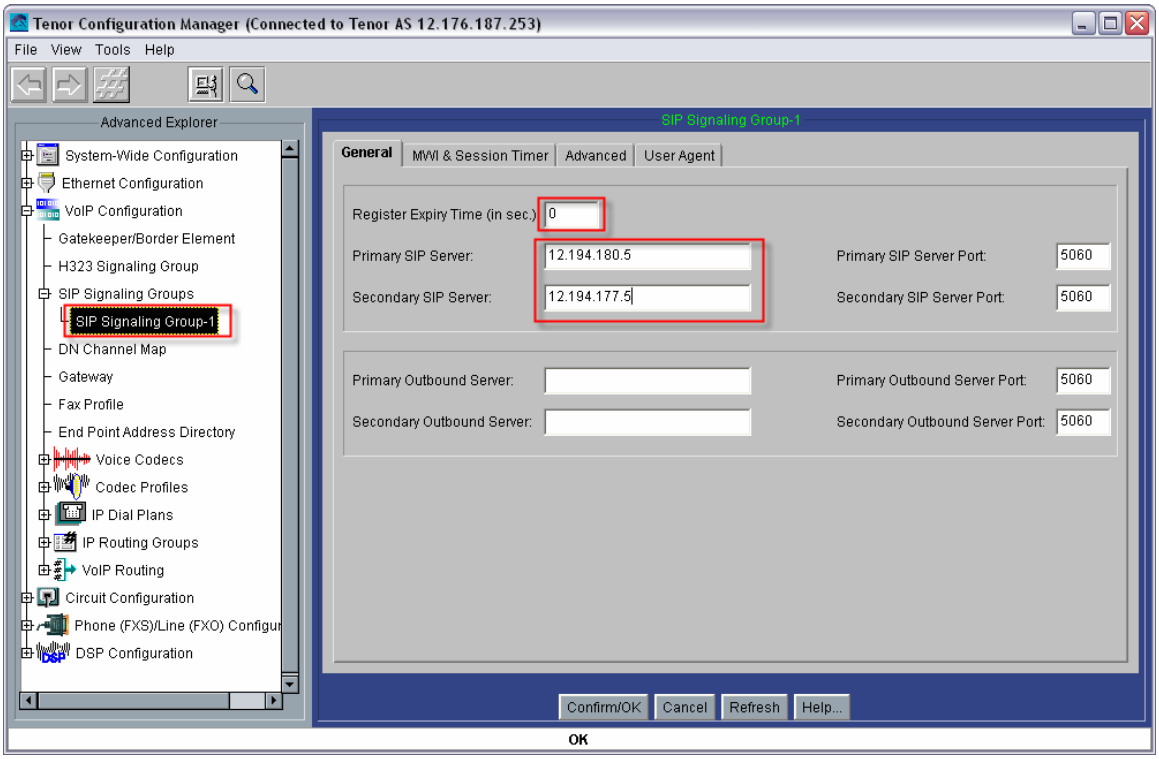
Step	Description
3.	Enter the Tenor **IP Address**, a **Description**, and the **Login ID** and **Password.** Click on **OK.** 

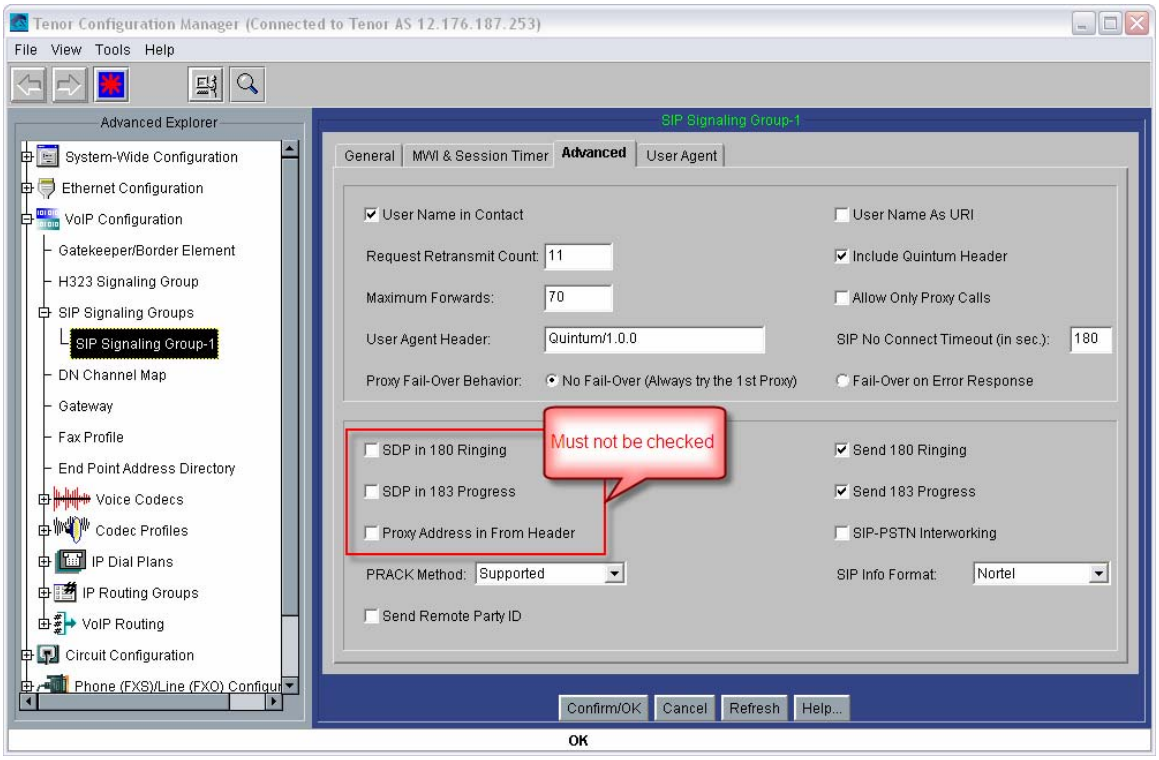
Step	Description
4.	Connect to the Tenor AS from the Tenor Configuration Manager. Highlight the Tenor AS switch and click on **Connect**. 

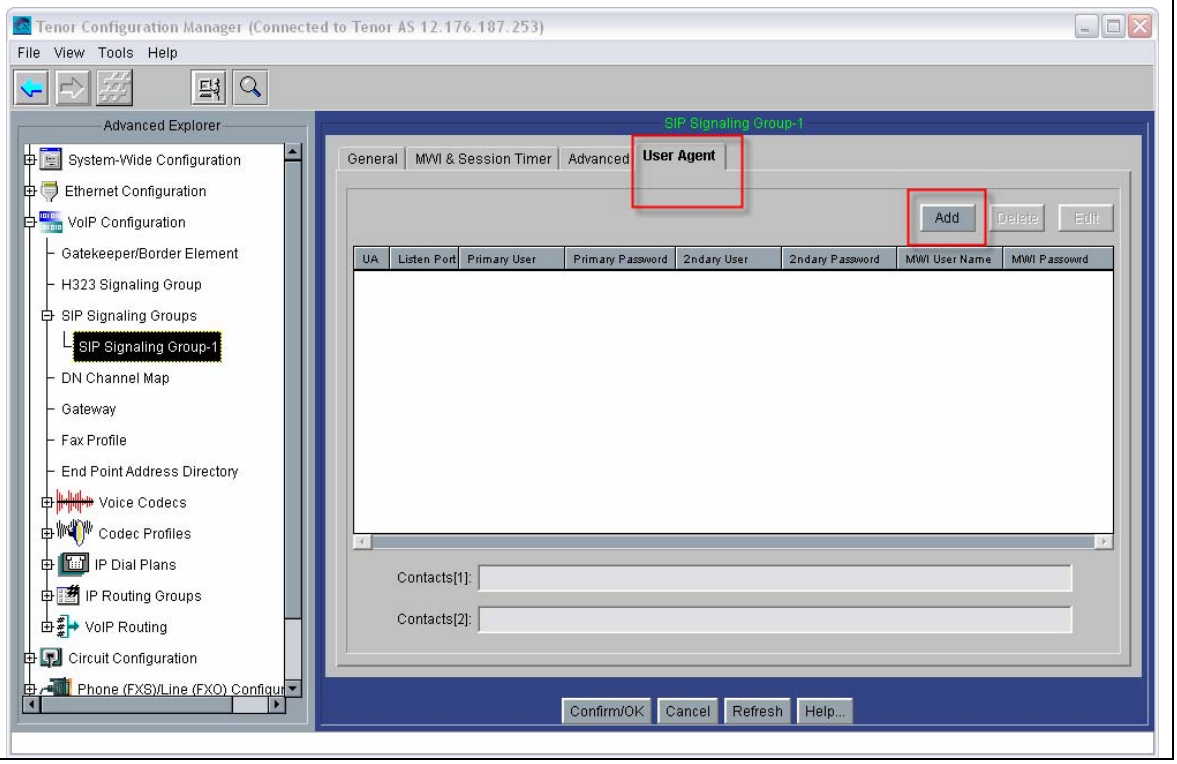
Step	Description
5.	Enter values for the Primary and Secondary DNS Server IP Address. If not using DNS enter: 0.0.0.0 Click **Confirm/OK** then the sunburst icon on the menu bar to implements the change. 

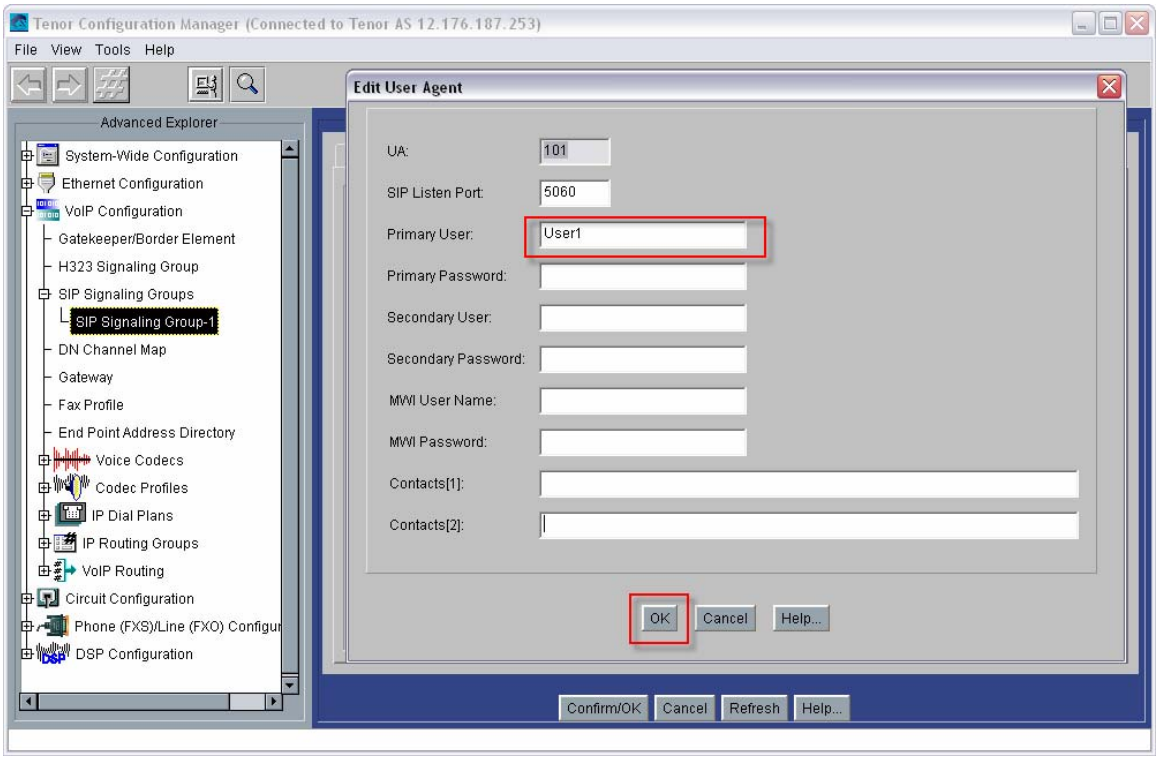
Step	Description
6.	Click on the **Advanced Explorer** icon on the menu bar. 

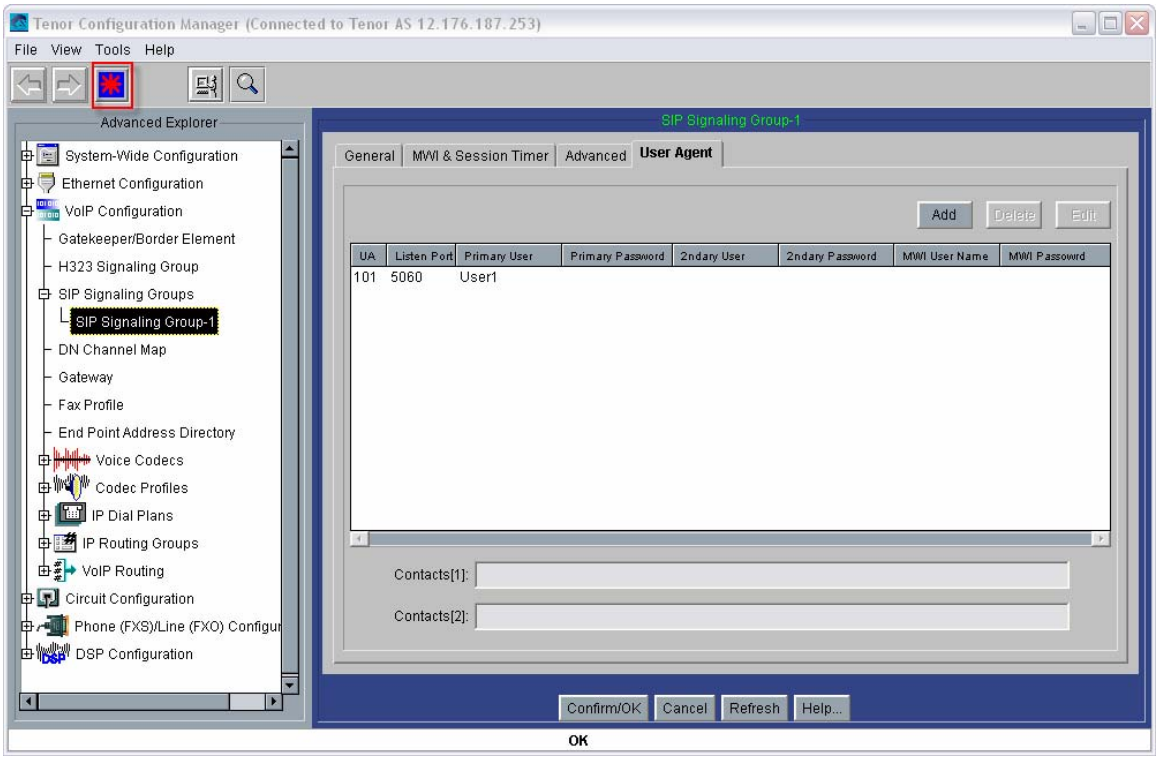
Step	Description
7.	From the **Advanced Explorer** panel on the left, highlight the **Dial Plan** field. Select the desired **Dial Plan Country** from the drop down menu. The sample configuration uses ***None***. Select the desired **Progress Tone Country** setting from the drop down menu. The sample configuration uses ***USA***. Enter values for the Minimum and Maximum dial digit string length. Click **Confirm/OK** then the sunburst icon on the menu bar to implements the change. 

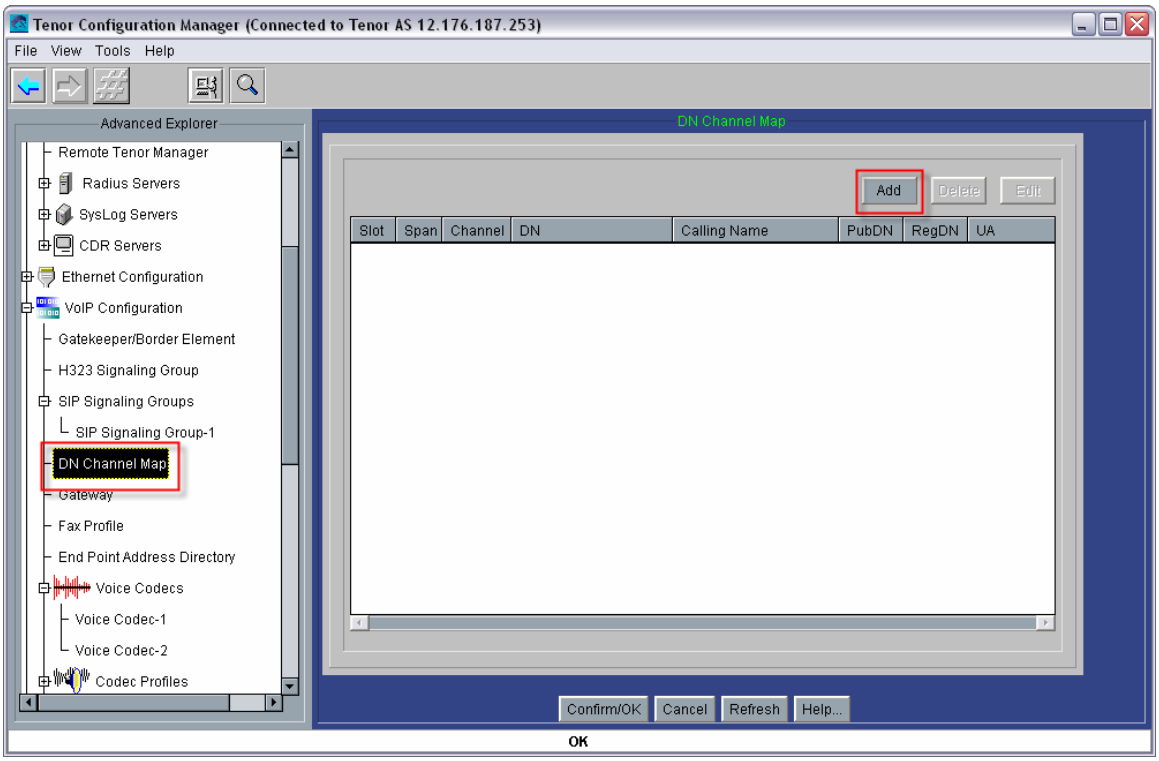
Step	Description
8.	From the **Advanced Explorer** panel on the left, click on the **+** sign next to **VoIP Configuration → SIP Signal Groups** to expand the field. Highlight the **SIP Signaling Group-1** field. Under the **General** tab, enter the **Primary SIP Server IP Address** and the **Secondary SIP Server IP Address** (IP Addresses of AT&T Primary and Secondary IP Border Elements). To disable Registration, enter the **Register Expiry Time** of **0.** 

Step	Description
9.	Click on the **Advanced** tab. Un-check the boxes for: • **"SDP in 180 Ringing"** • **"SDP in 183 Progress"** • **"Proxy Address in From Header"** 

Step	Description
10.	Click on the **User Agent** tab. Click the **Add** button to add display the Add User Agent pop-up window. 

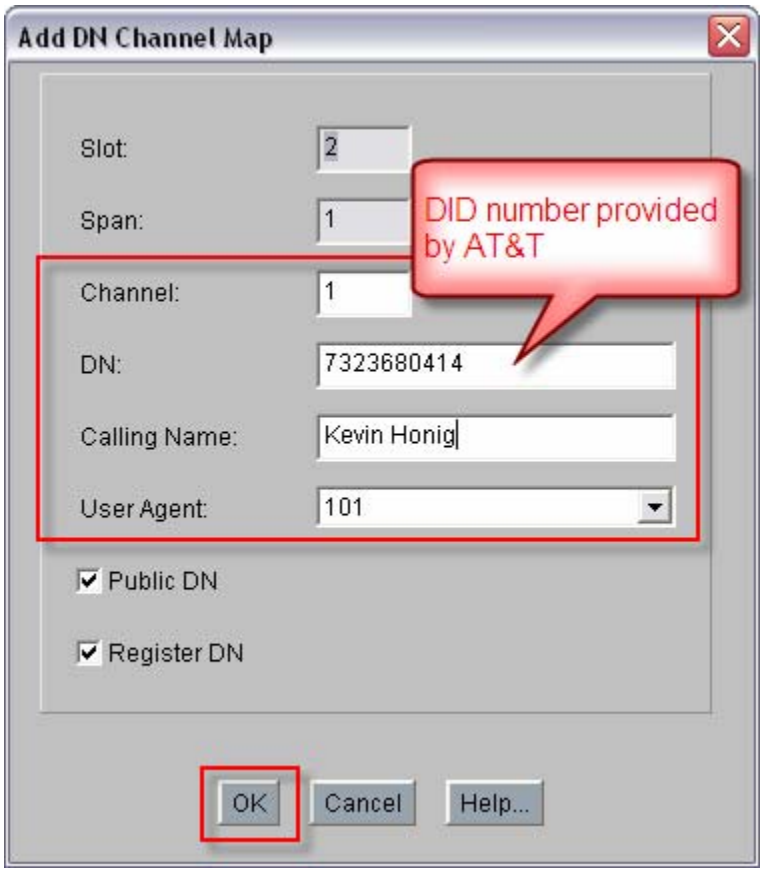
Step	Description
11.	In the **Add User Agent** pop-up window, enter the following information: **PrimaryUser** - The username for Registration and Authentication purposes. If Registration were enabled, the "username" will appear in the URI populated in the To and From headers of the REGISTER message. **Primary User:** ***User1*** < --- Any alpha-numeric string may be entered because SIP Registration and Authentication are not applicable to the AT&T IP Flexible Reach service. Click **OK** to continue. 

Step	Description
12.	At the **SIP Signal Group-1** panel click **Confirm/OK** to complete and the sunburst icon to implement the change in the Tenor AS. 

Step	Description
13.	From the **Advanced Explorer** panel on the left, highlight the **DN Channel Map** field. Click **Add** on the **DN Channel Map** panel on the right. 

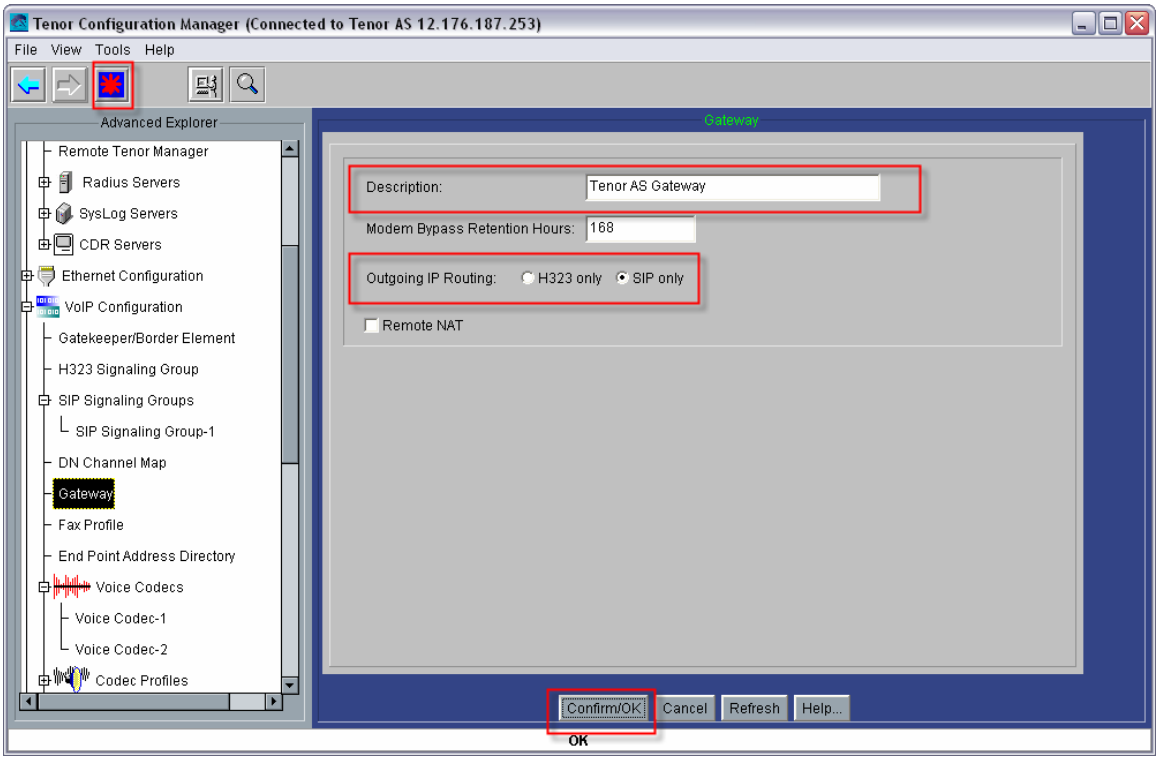
14. At the **Add DN Channel Map** pop-up window, enter the following information.

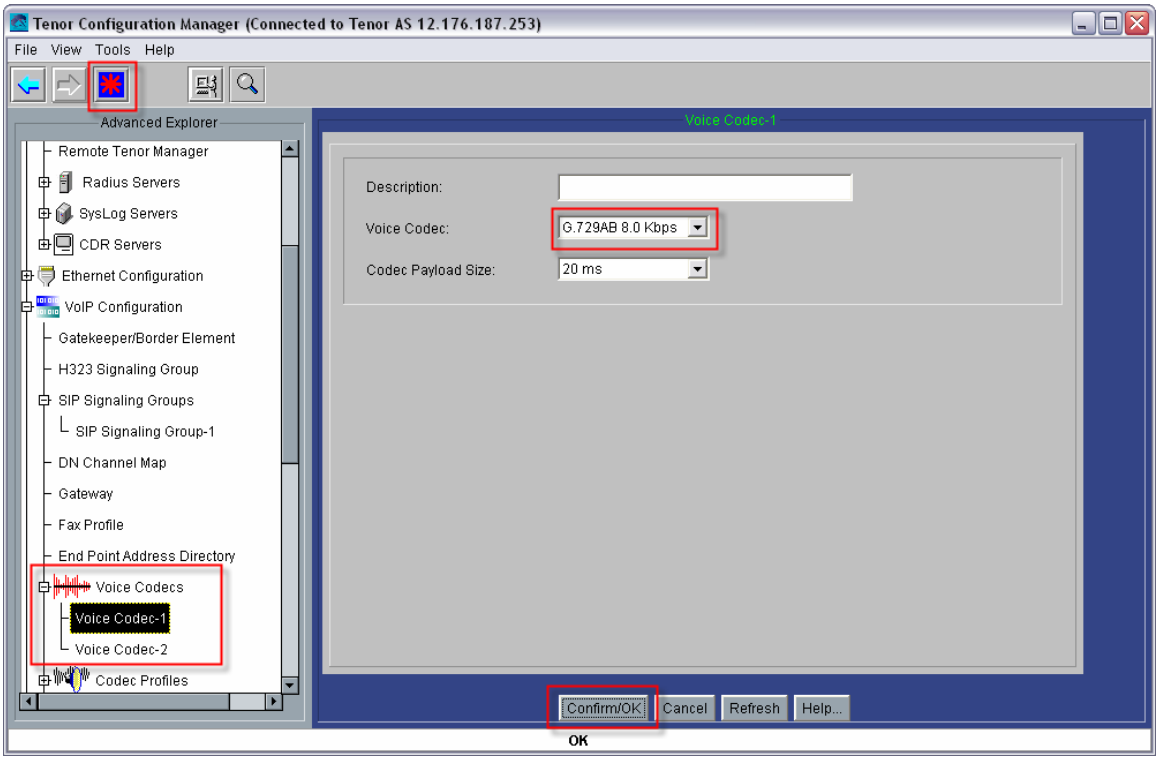
Channel: **1** < --- Physical port used on Tenor
DN: **7323680414** < --- Phone Number (DID number provided by AT&T)
Calling Name: **Kevin Honig** < --- Display Name
User Agent: ***101*** < --- User Agent defined in Step 11.
Public DN **checked** < --- default
Register DN **checked** < --- default

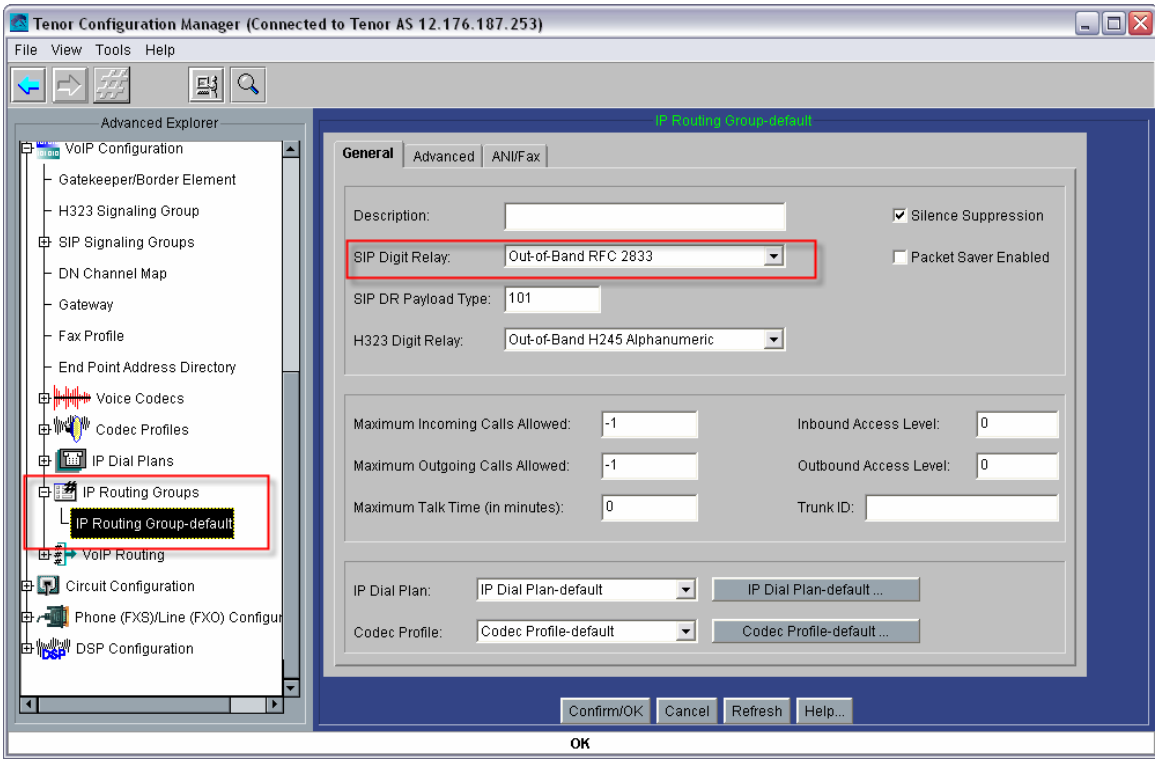


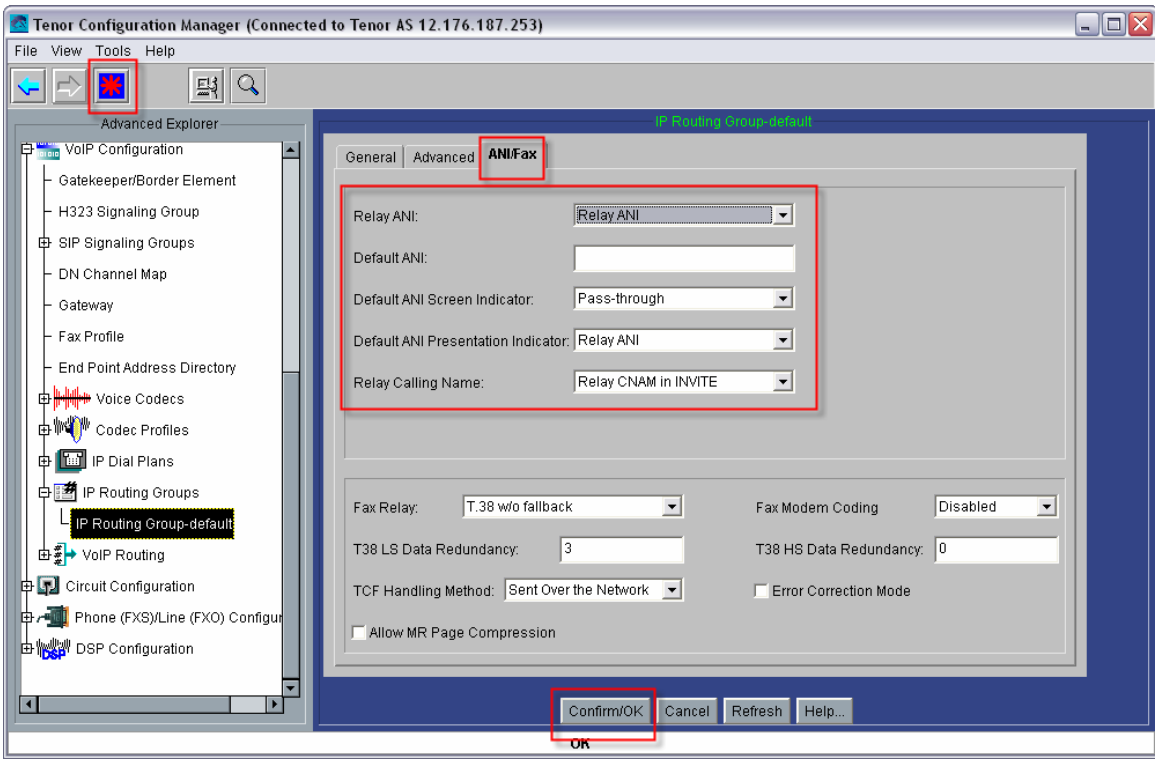
Click **OK** to continue. At the DN Channel Map panel click **Confirm/OK** and the sunburst icon implements the change.

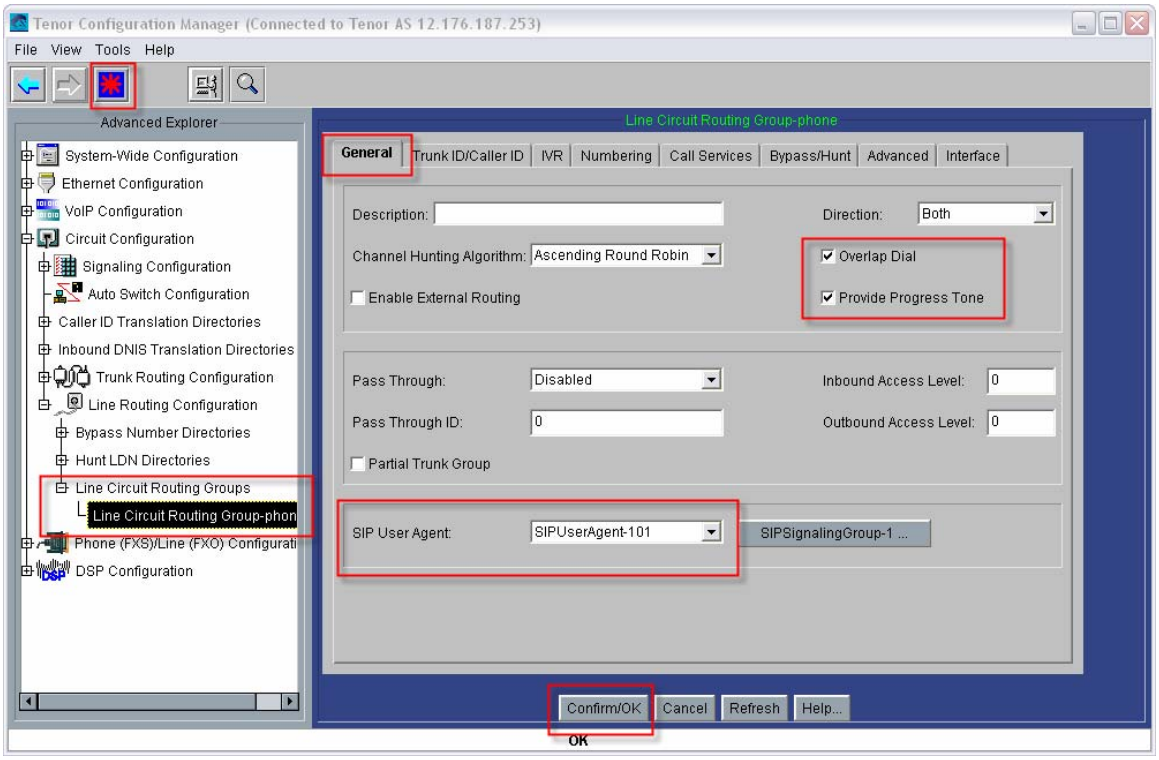
Note:
Slot *and* ***Span*** *are not relevant to the Analog Tenor.*
Channel: *Denotes the physical port that the analog device will be connected.*
DN: *DID number provided by AT&T. Populated in outgoing INVITE message (to AT&T) as the user part of the URI in the From and Contact headers. On inbound calls to*

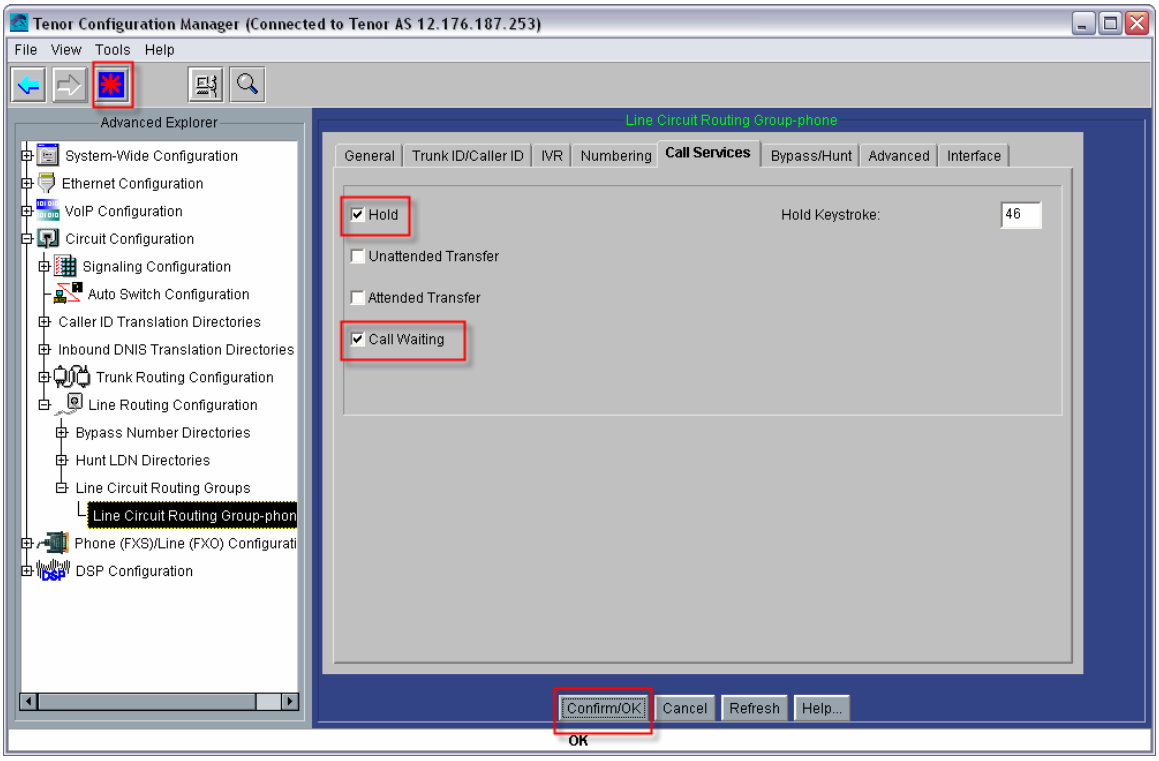
	Tenor, used to determine routing of calls to physical line. Should appear as user part of Request URI of incoming INVITE. ***Calling Name:*** *Will appear as the Display Name in the From header in outgoing INVITE messages.* ***Public DN:*** *Indicates whether or not this is a Public DN* ***Register DN:*** *Only relevant to H.323*
15.	From the **Advanced Explorer** panel on the left, highlight the **Gateway**. Enter a **Description** and ***check*** the **SIP only** radio button for the **Outgoing IP Routing** field under the Gateway screen panel on the right. Click **Confirm/OK** then the sunburst icon on the menu bar to implements the change. 

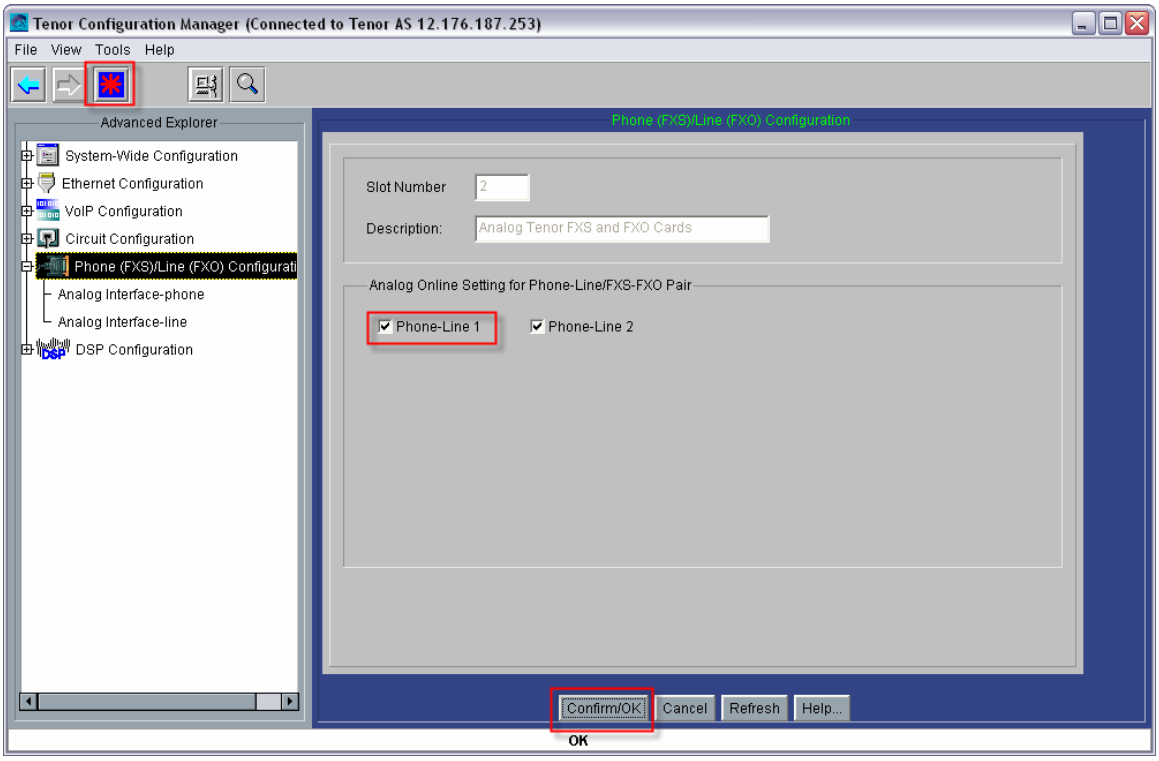
16.	From the **Advanced Explorer** panel on the left, click on the **+** sign to expand the **Voice Codecs** field. Highlight the **Voice Codec-1** field. Select the desire **Voice Codec** field from the drop down menu. The sample configuration uses the ***G.729*** codec. Click **Confirm/OK** then the sunburst icon on the menu bar to implements the change. 

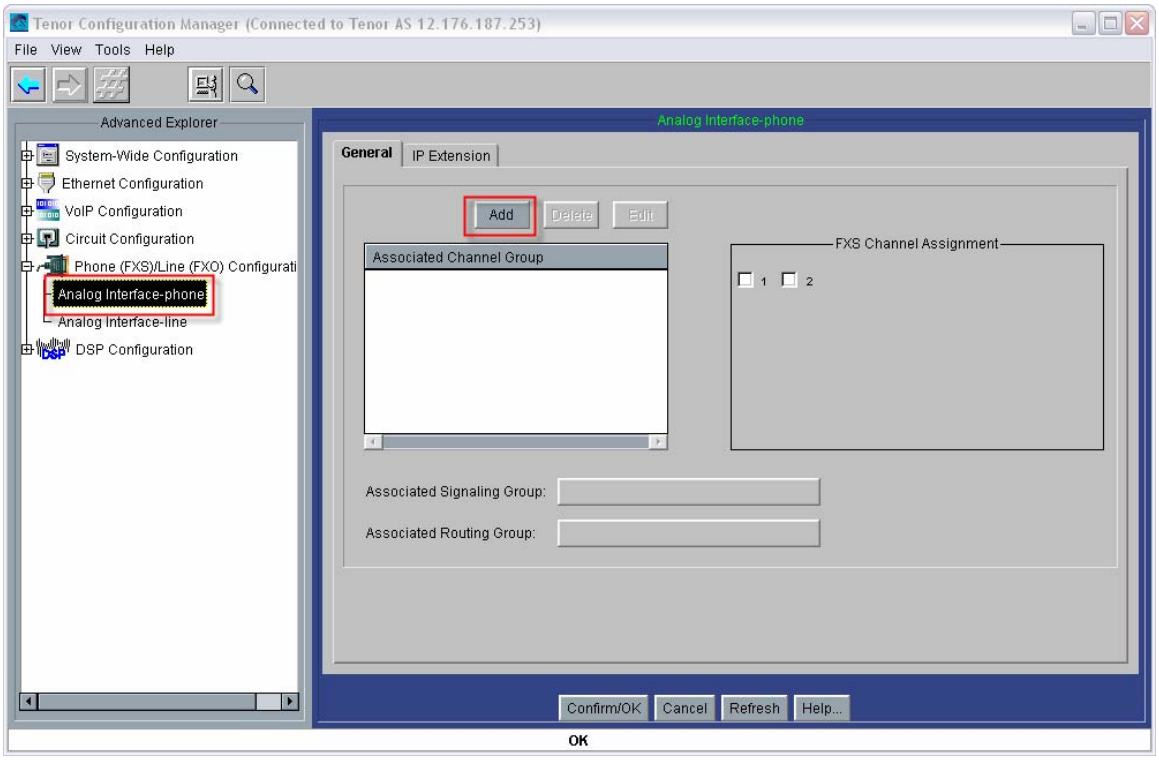
17.	From the **Advanced Explorer** panel on the left, highlight the **IP Routing Group-default** field under **IP Routing Groups**. Under the **General** tab in the **IP Routing Group-default** panel on the right, select ***Out-of-Band RFC 2833*** for **SIP Digit Relay** from the drop down menu. 

18.	Click on the **ANI/FAX** tab under the **IP Routing Group-default** panel on the right. Select ***Relay ANI*** for **Relay ANI** from the drop down menu. Select ***Pass-through*** for **Default ANI Screen Indicator** from the drop down menu. Select ***Relay ANI*** for **Default ANI Presentation Indicator** from the drop down menu. Select ***Relay CNAM in INVITE*** for **Relay Calling Name** from the drop down menu. Click **Confirm/OK** then the sunburst icon on the menu bar to implements the change. 

19.	From the **Advanced Explorer** panel on the left, expand **Circuit Configuration → Line Routing Configuration → Line Circuit Routing Groups**, and highlight the **Line Circuit Routing Group-phone** field. Click on the **General** tab under the **Line Circuit Routing Group-phone** panel on the right. From the **SIP User Agent** drop down menu, select ***SIPUserAgent-101*** and ***check*** the boxes for **Overlap Dial** and **Provide Progress Tone**. Click **Confirm/OK** then the sunburst icon on the menu bar to implements the change. Click the **Call Services** tab. 

20.	From the **Call Services** tab under the **Line Circuit Routing Group-phone** panel on the right. Check to enable **Hold**, and **Call Waiting**. Disable **Unattended Transfer** and **Attended** Transfer if checked. Click **Confirm/OK** then the sunburst icon on the menu bar to implements the change. Click on **ByPass/Hunt** tab to continue. 

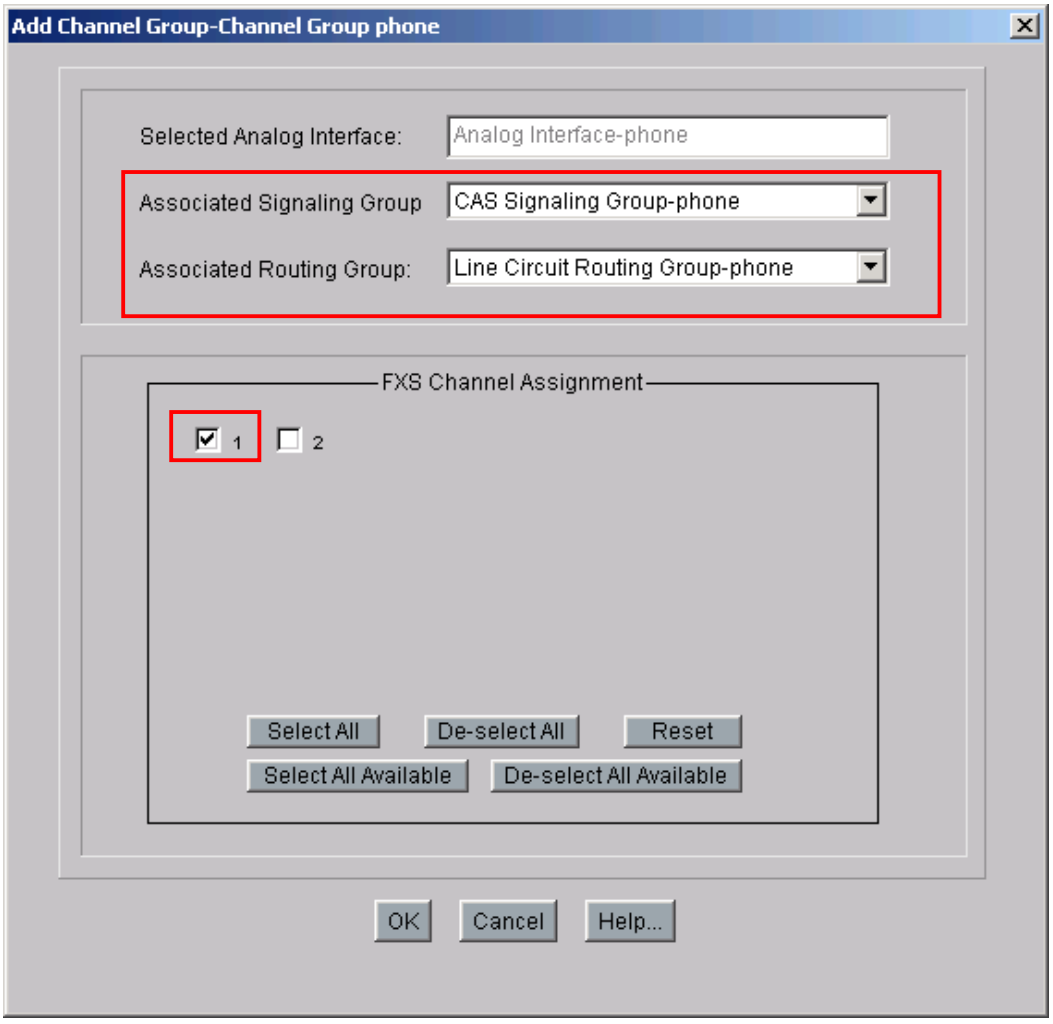
21.	Under the **Advanced Explorer** panel on the left, highlight the **Phone (FXS)/Line (FXO) Configuration**. Check the box to enable **Phone-Line 1**. Click **Confirm/OK** then the sunburst icon on the menu bar to implements the change. 

22.	Under the **Advanced Explorer** panel on the left, expand **Phone (FXS)/Line (FXO) Configuration**, and highlight the **Analog interface-phone** field. Highlight **Channel Group-phone** then click **Add**. 
23.	Enter a description for the **Channel Group** and click **OK** to continue.

24. In the **Add Channel Group-Channel Group phone** pop-up window, select the following information.

Associated Signaling Group:	***CAS Signaling Group-phone***
Associated Routing Group:	***Line Circuit Routing Group-phone***
FXS Channel Assignment	***Check*** radio button for **1**

Click **OK** to complete.

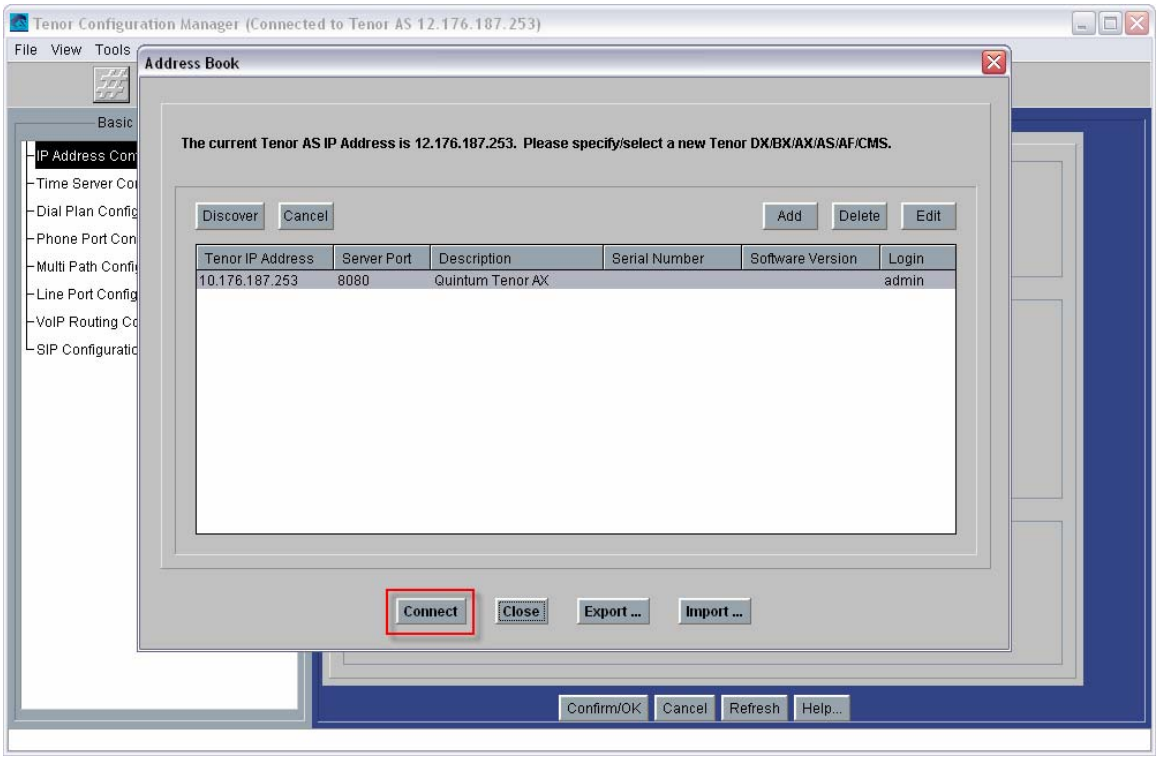


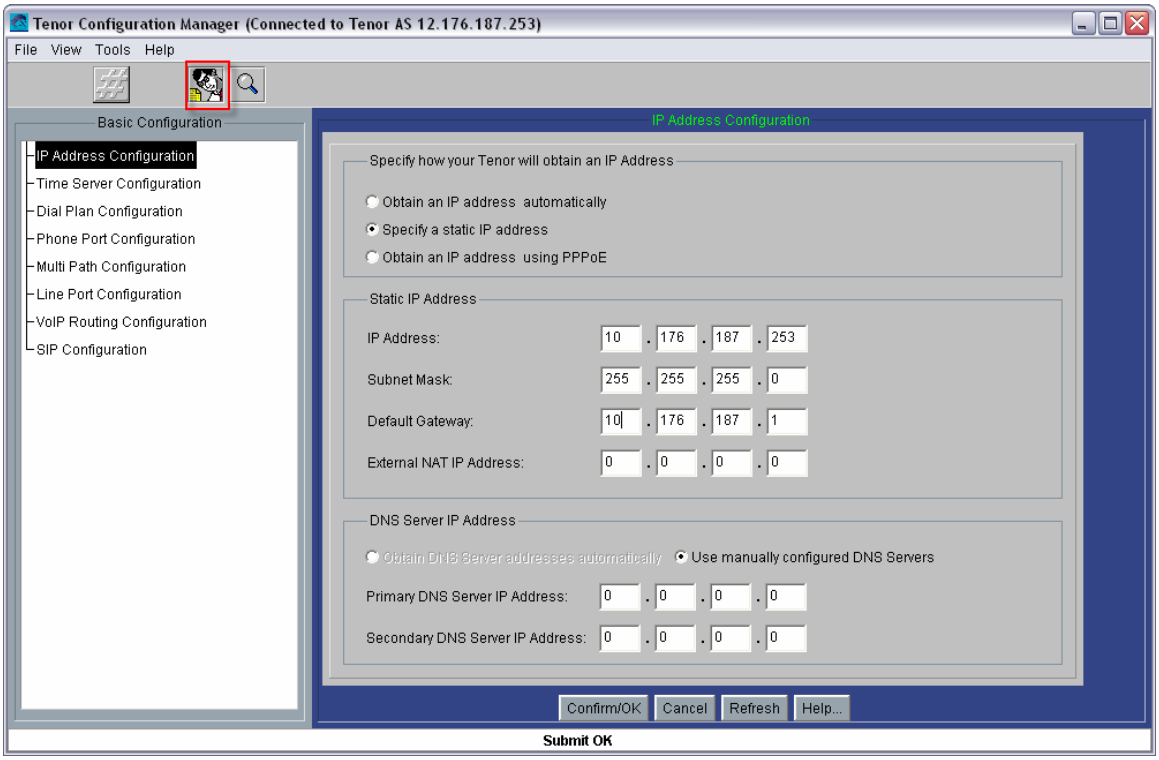
5.3 Configuration for Super G3 FAX

When there are Super G3 FAX machines on both ends of the call, the Tenor does not detect "fax" and never switches from voice to T.38. This causes the FAX call to fail. This issue can be circumvented by configuring the Tenor to use G.711 exclusively. Unfortunately G.729 is the preferred codec for voice calls. On the Tenor it is not possible to configure one analog line for G.711 and the remaining lines for G.729. All lines on the Tenor use the same default codec values.

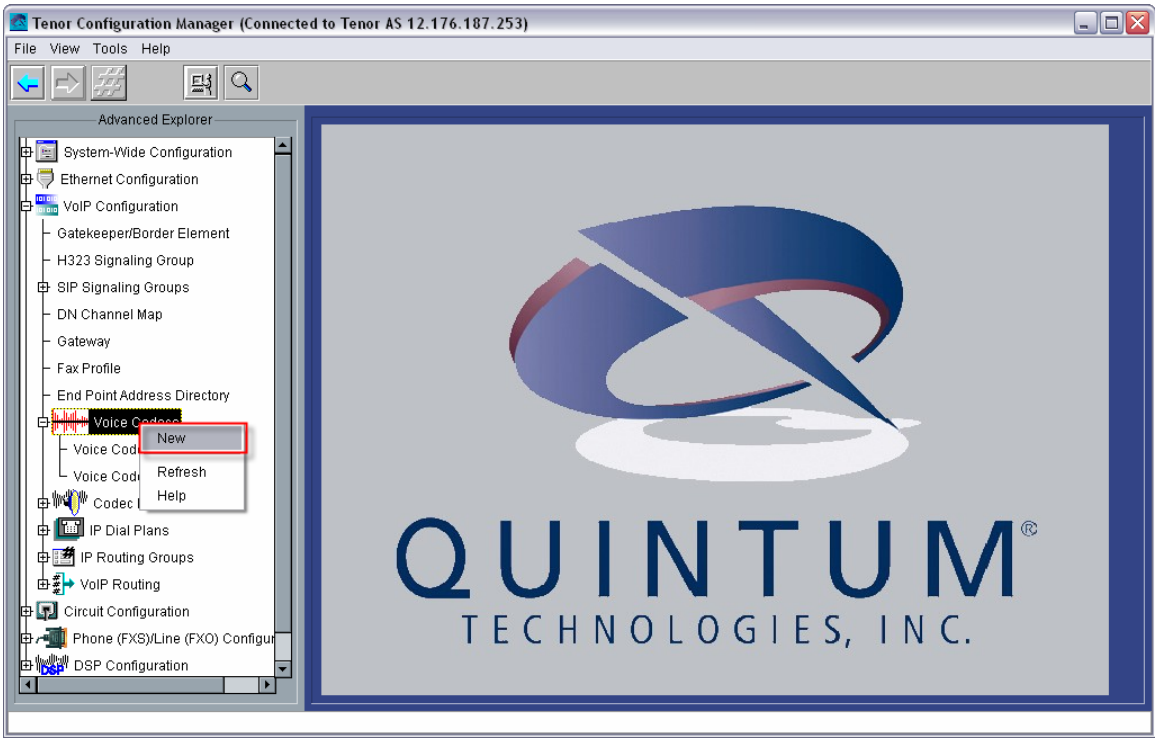
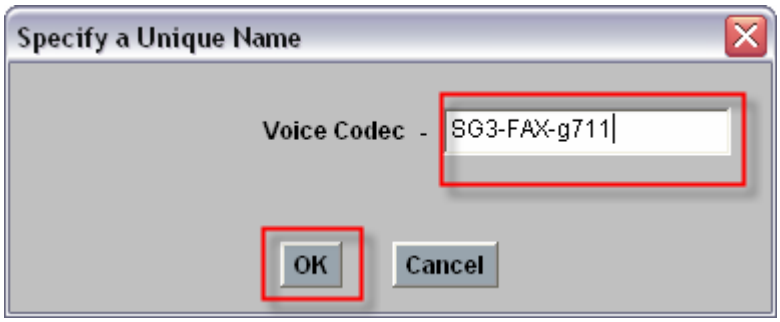
To help avoid the failure of outbound Super G3 FAX calls, it is possible to configure a static route for specific telephone numbers or number patterns and associate the G.711 Codec with those numbers. So, if the customer has a SG3 fax machine attached to a Tenor and makes calls to other SG3 machines at specific numbers, they can configure G.711 for just those numbers. This does not work for incoming calls.

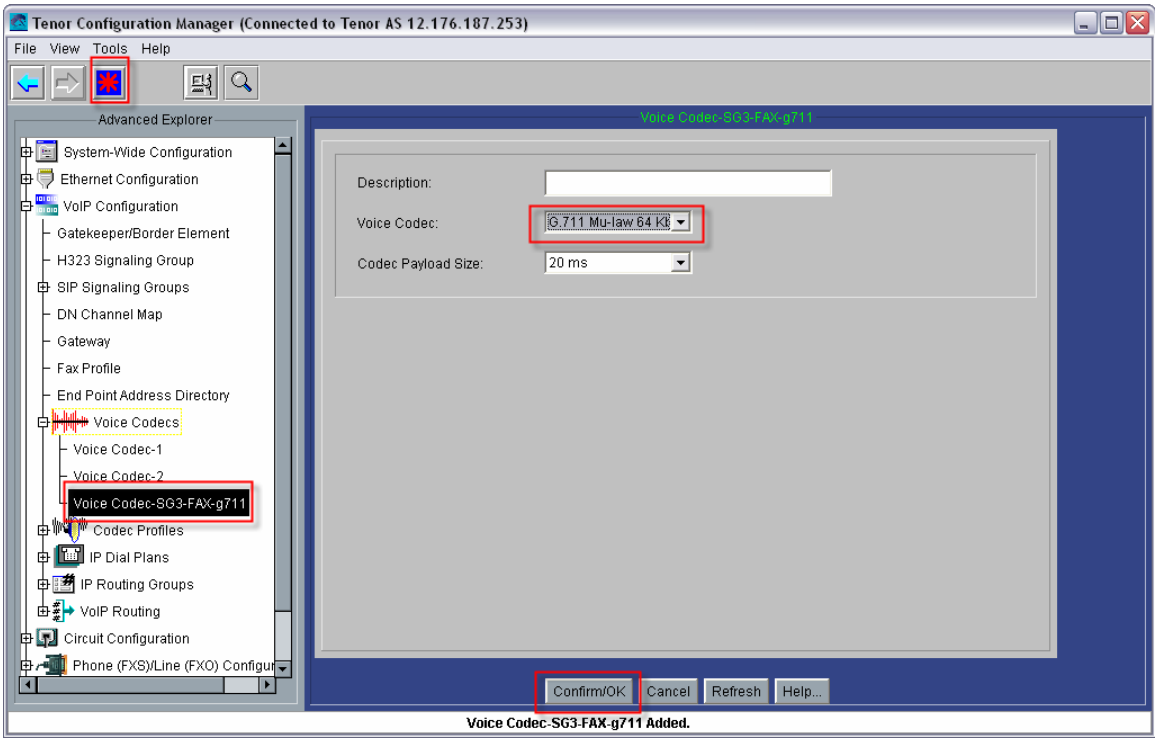
The following steps describe the additional configuration necessary for the Tenor to use a Default Route with a G.711 Codec for known SG3 machines.

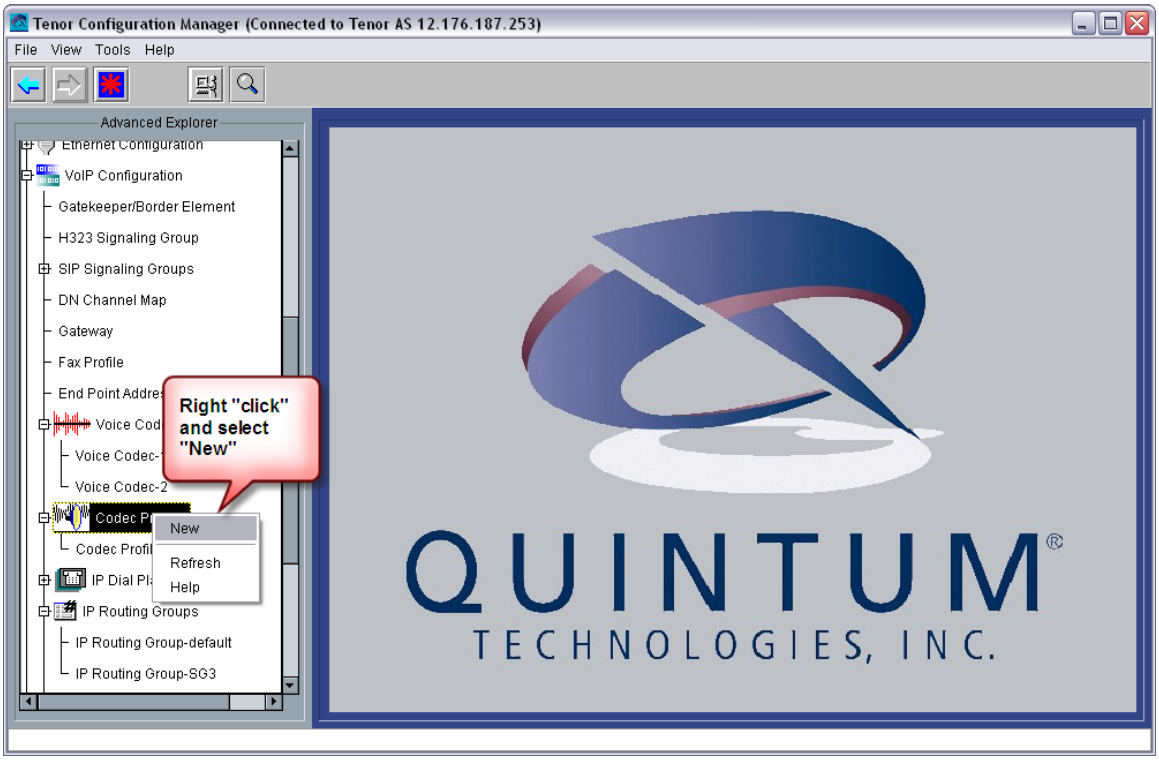
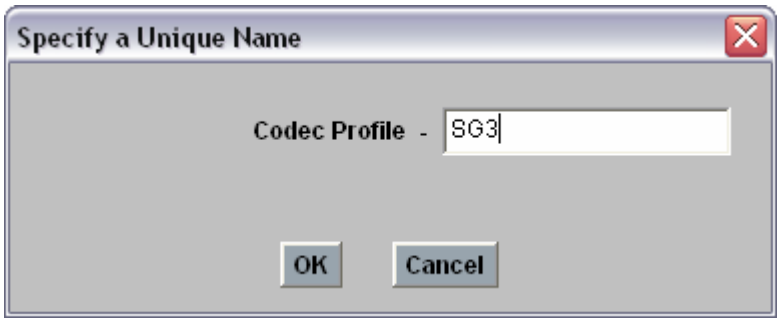
Step	Description
1.	Connect to the Tenor AS from the Tenor Configuration Manager. Highlight the Tenor AS switch and click on **Connect**. 

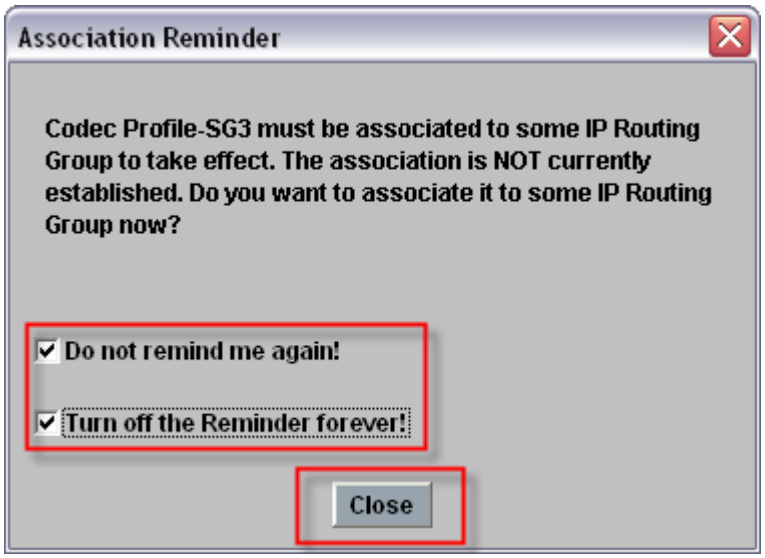
Step	Description
2.	Click on the **Advanced Explorer** icon on the menu bar. 

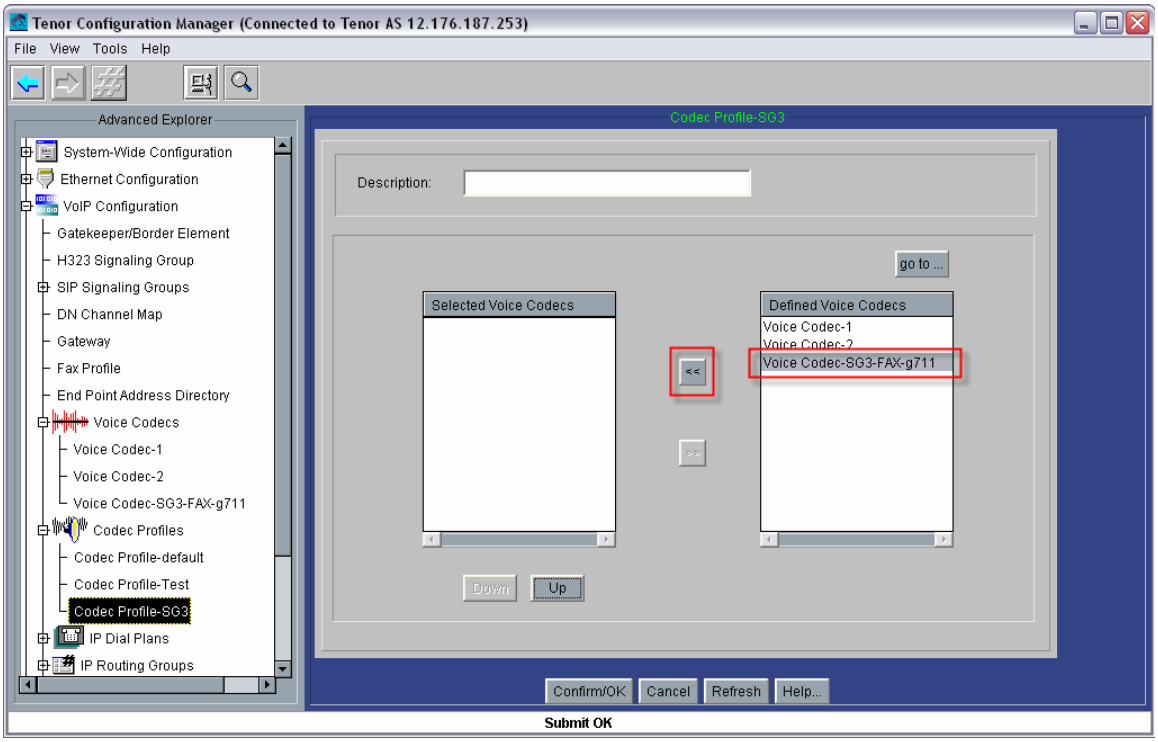
Tenor Configuration Manager (Connected to Tenor AS 12.176.187.253)
File View Tools Help
Basic Configuration
IP Address Configuration
Time Server Configuration
Dial Plan Configuration
Phone Port Configuration
Multi Path Configuration
Line Port Configuration
VoIP Routing Configuration
SIP Configuration
IP Address Configuration
Specify how your Tenor will obtain an IP Address
Obtain an IP address automatically
Specify a static IP address
Obtain an IP address using PPPoE
Static IP Address
IP Address: 10 . 176 . 187 . 253
Subnet Mask: 255 . 255 . 255 . 0
Default Gateway: 10 . 176 . 187 . 1
External NAT IP Address: 0 . 0 . 0 . 0
DNS Server IP Address
Obtain DNS Server addresses automatically
Use manually configured DNS Servers
Primary DNS Server IP Address: 0 . 0 . 0 . 0
Secondary DNS Server IP Address: 0 . 0 . 0 . 0
Confirm/OK
Cancel
Refresh
Help...
Submit OK

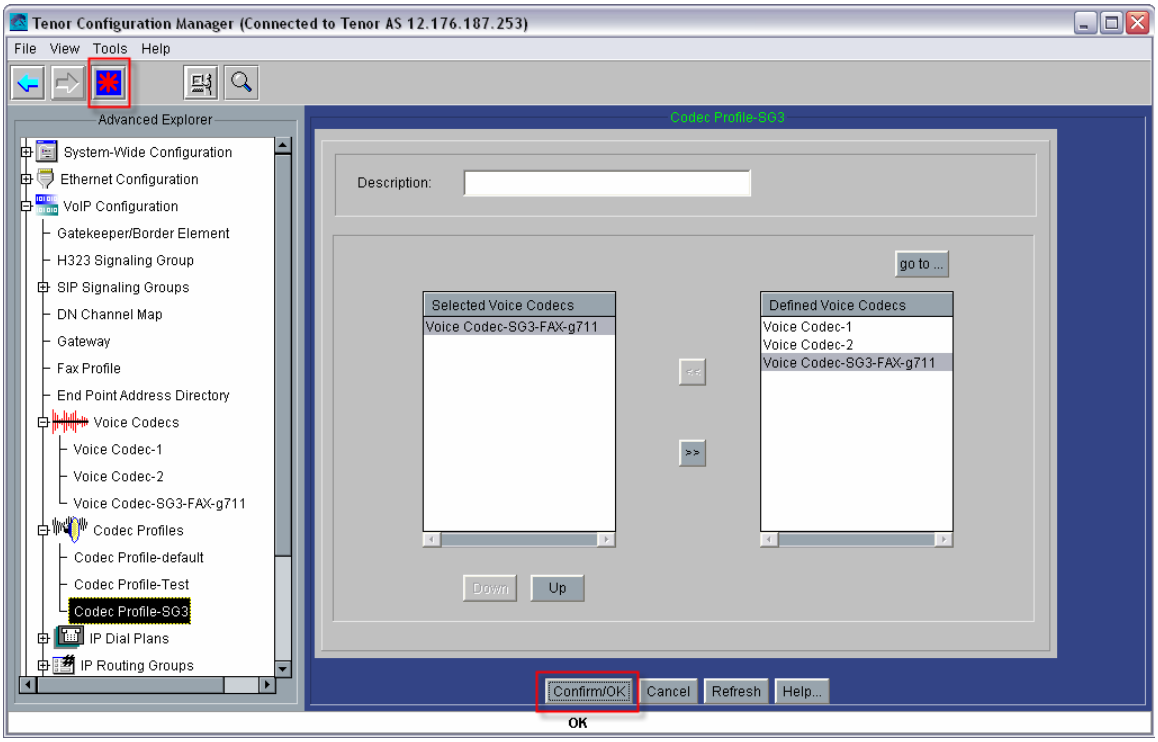
Step	Description
3.	Under the **Advanced Explorer** panel on the left, highlight the **Voice Codecs** entry. Right Check and select "**New"** from the command directory that appears. 
4.	At the **Add Voice Codec** pop-up window, enter a name for the new codec (SG3-FAX-g711). Click **OK** to continue. 

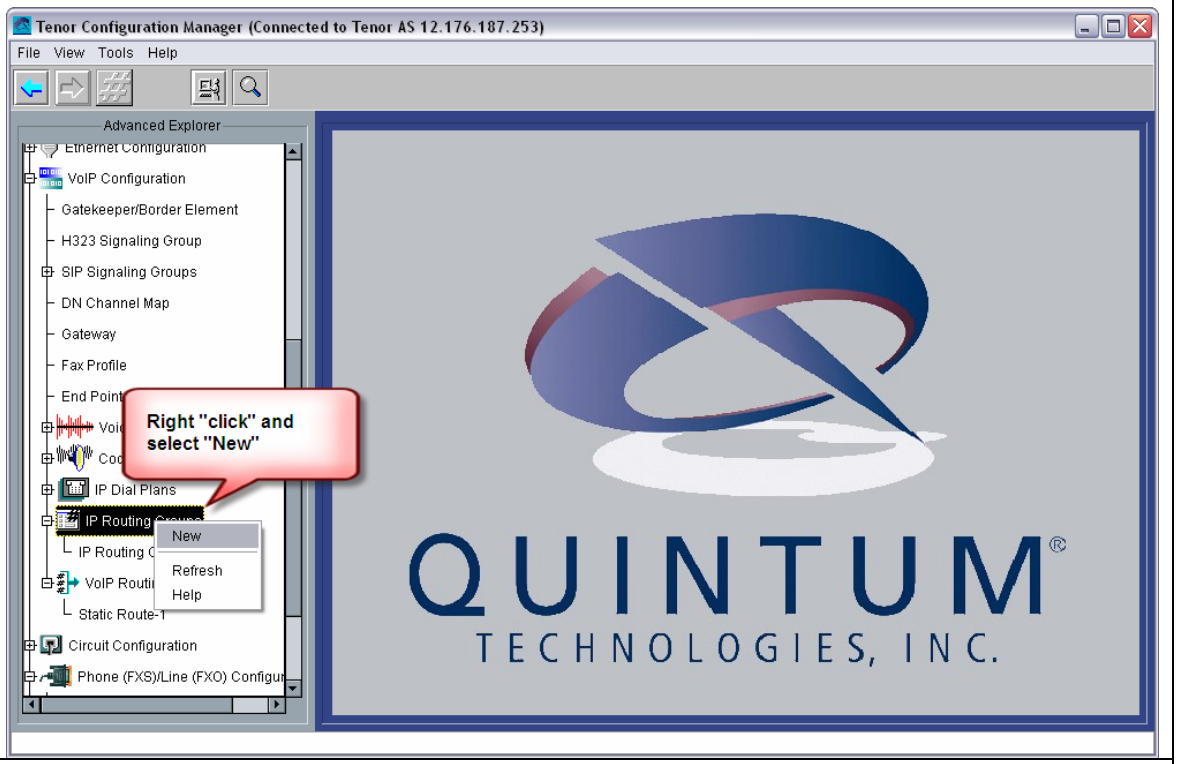
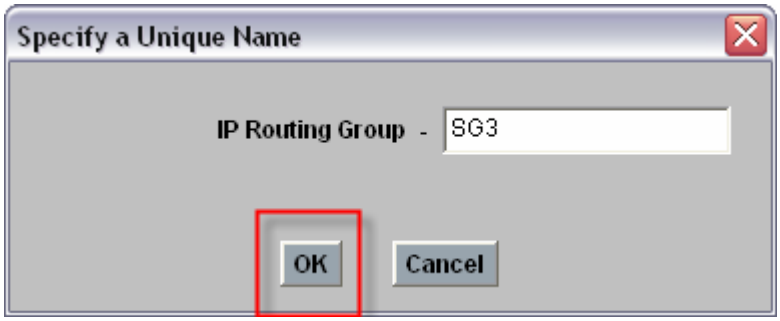
Step	Description
5.	From the **Advanced Explorer** panel on the left, highlight the **Voice Codec-SG3-FAX-g711** field. Select the desired **Voice Codec** field from the drop down menu (***G.711 Mu-law*** codec).  Click **Confirm/OK** then the sunburst icon on the menu bar to implements the change.

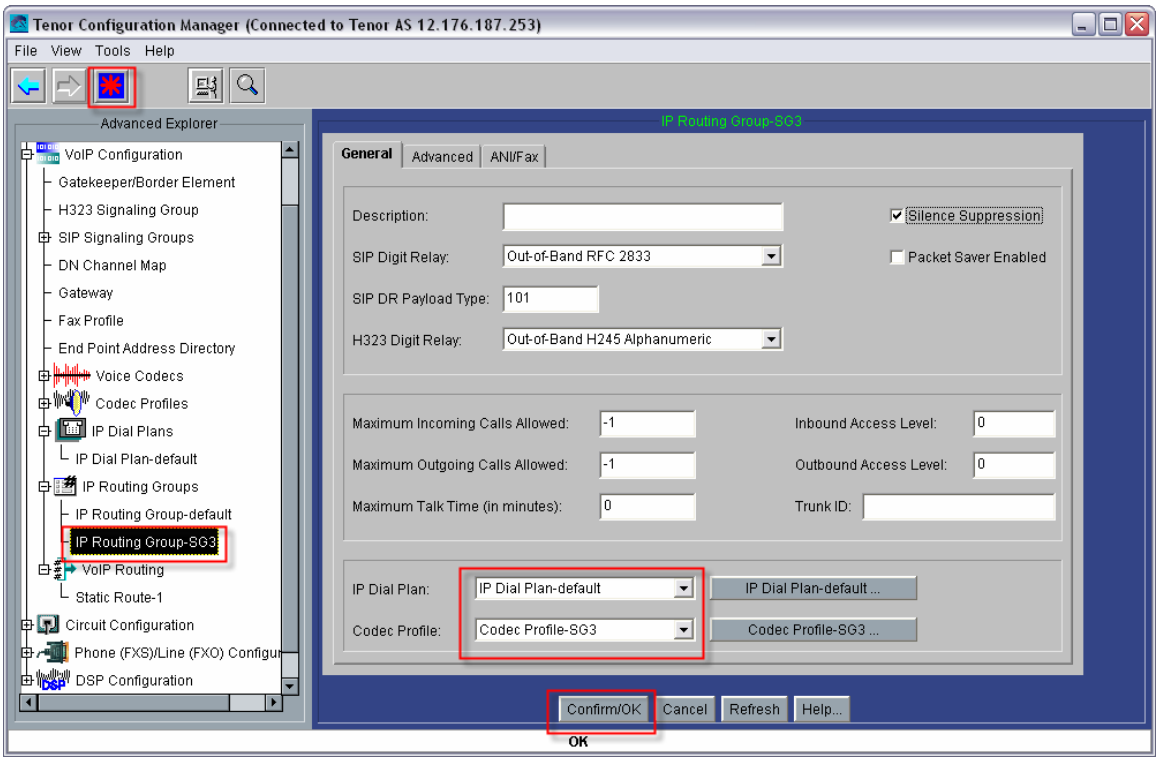
Step	Description
6.	Under the **Advanced Explorer** panel on the left, highlight the **Codec Profiles** entry. Right Check and select "**New**" from the command directory that appears. 
7.	At the **Add Codec Profile** pop-up window, enter a name for the new profile (SG3). Click **OK** to continue. 

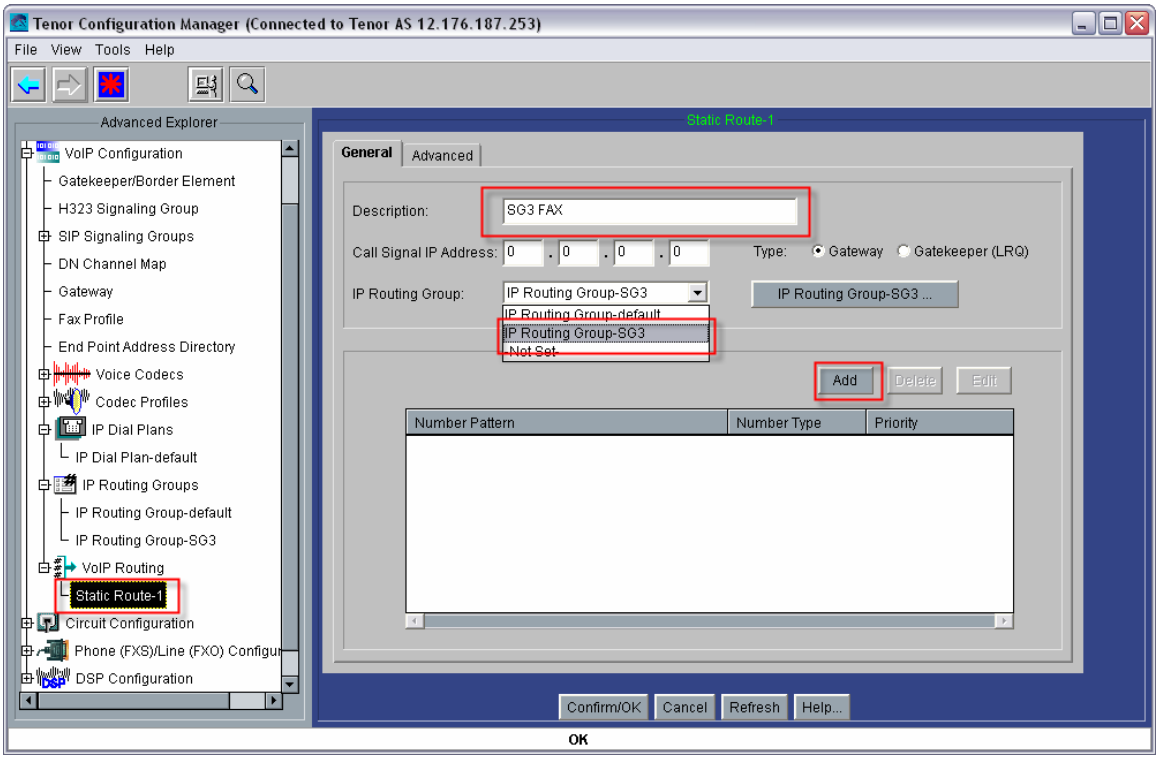
Step	Description
8.	If an ***"Association Reminder"*** appears, check "Do not remind me again!" and "Turn Off the Reminder forever!". Click **Close** to continue. 

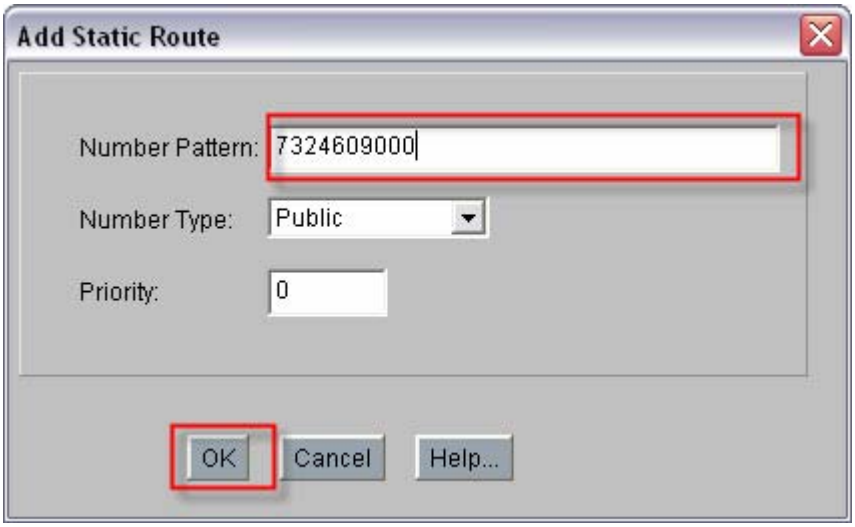
Step	Description
9.	In the **Defined Voice Codecs** window, highlight the **Voice Codec** defined as **g711**. **Click** the **"<<"** button to move the g711 Voice Codec to the **Selected Voice Codec** window. 

Step	Description
10.	Click **Confirm/OK** then the sunburst icon on the menu bar to implements the change. 

Step	Description
11.	Under the **Advanced Explorer** panel on the left, highlight the **IP Routing Group** entry. Right Check and select "**New**" from the command directory that appears. 
12.	At the **Add IP Routing Group Profile** pop-up window, enter a name for the new profile (SG3). Click **OK** to continue. 

Step	Description
13.	From the **Advanced Explorer** panel on the left, click on the **+** sign next to **VoIP Configuration → IP Routing Groups** to expand the field. Highlight the **IP Routing Group SG3** field. Under the **General** tab, select the desired **Codec Profile (SG3)** and the **IP Dial Plan (default)** from the drop down menus Click **Confirm/OK** then the sunburst icon on the menu bar to implements the change. 

Step	Description
14.	From the **Advanced Explorer** panel on the left, click on the **+** sign to expand **the VoIP Routing** field. Highlight the **Static Route-1** field. Enter a **Description** and Select the **IP Routing Group** configured for Super G3 FAX from the drop down menu. Click **Add** button above the **Number Pattern** panel on the right. 

Step	Description
15.	At the **Add Static Route** pop-up window, enter the phone number for the Super G3 Fax Machine.  Click **OK** to continue. At the Static Route-1 panel click **Confirm/OK** and the sunburst icon implements the change.

6 Troubleshooting

For technical support on the Quintum Tenor AS and Tenor AX, contact Quintum at 877-435-7553, and also refer to www.quintum.com

7 Additional References

[1] Tenor AX VoIP Multipath/Gateway Switch Product Guide, P/N 480-0062-00-12
http://www.quintum.com/support/products/2G/tenor_2G/sysdoc/Tenoraxuserguide.pdf

[2] Tenor AS VoIP Multipath/Gateway Switch Product Guide, P/N 480-0059-00-13
http://www.quintum.com/support/products/2G/tenor_2G/sysdoc/TenorASuserguide.pdf

[3] Tenor Configuration Manager/Tenor Monitor Product Guide. P/N 480-0028-00-05
http://www.quintum.com/support/mgmt/TenorConfigManagerUsersGuide.pdf

Appendix 1 Dump of Tenor Database Configuration

The following Word document contains a complete printout of the Tenor database configuration used for testing with the AT&T IP Flexible Reach Service. Click here to download.